USER MANUAL
Configuration Program

Version 1.6

Dear customer,

Our compliments on buying the ***2N*** **- OMEGA Lite***. This new product was developed with an emphasis on the maximum possible use value, quality and reliability. We hope that you will be utterly satisfied with the 2N* - OMEGA Lite *for many years to come.*

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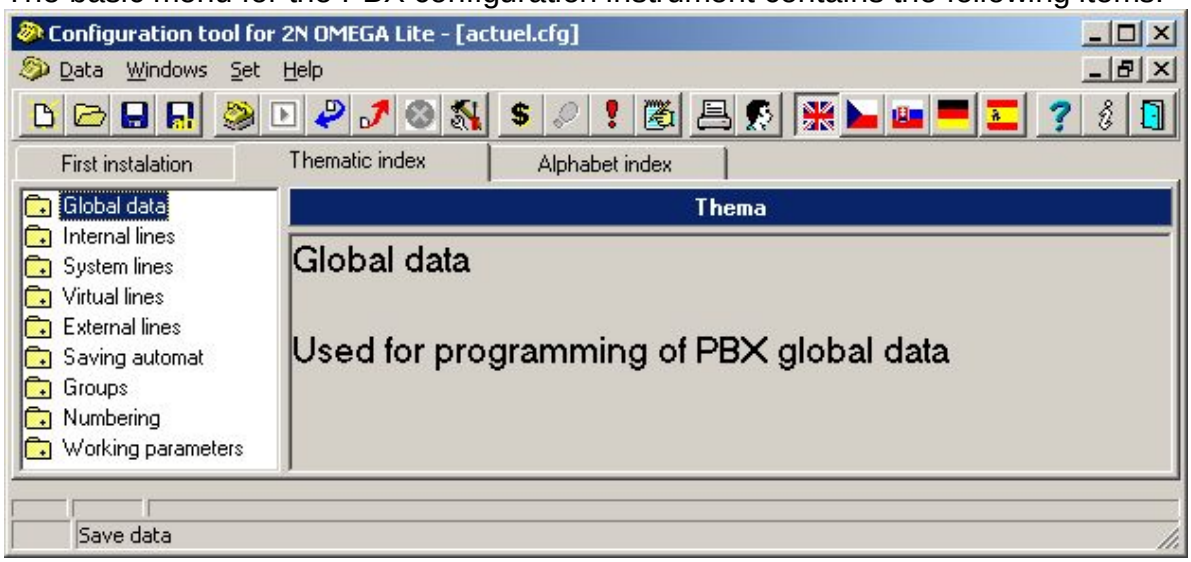
1. The PBX configuration instrument menu

The basic menu for the PBX configuration instrument contains the following items:



1.1. Data

- New — Making a new PBX configuration file
- Open… — Opening a current configuration file
- Close — Closing an open window with a configuration file
- Save — Saving a file
- Save as … — Saving a file under another name

- Firm values — Using the factory settings in the default set up

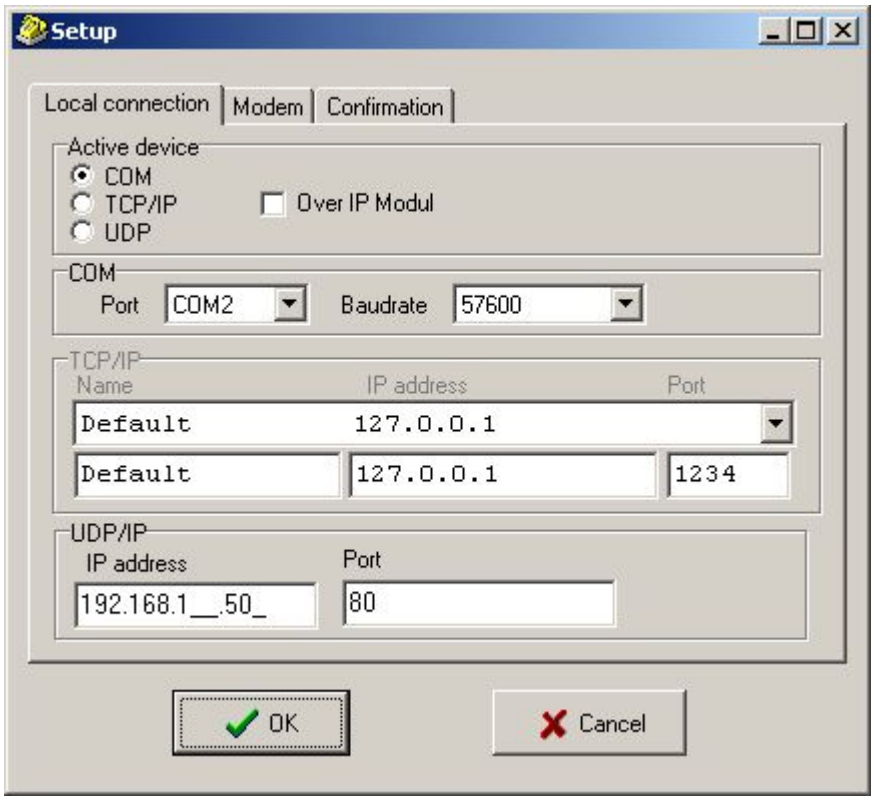
- From the PBX — Reading the configuration from the PBX
- To the PBX — Saving a configuration into the PBX
- Reset the PBX — Reset the PBX (the same function as turning on and off)
- Time synchronization — Setting the time to the PC which is connected to the PBX
- Upload the PBX software — Loading firmware into Flash.
- Reset and boot the PBX from Flash — Reset the PBX with firmware support into Flash and rewriting it into RAM.
- Freezing the PBX — The PBX switches to the service regime. Ongoing calls are completed other calls cannot be started. This state is indicated by the sign “PBX repairs!” on the system phones. Afterwards it is possible to turn the PBX off and change the modules. The PBX switches to the operating state after the command “Unfreeze PBX” or after turning on or off.
- Unfreezing the PBX — Returning to the operating state.
- Print actuel window
- Printing sets
- Print setup
- Close the program

1.2. Windows

- Cscade Grouping the windows into a cascade
- Tile Grouping the windows next to each other without overlap
- Arrange icons

- Panel The state of the lines
- Account data Displaying calls
- PBX diagnostics Displaying log system reports

1.3. Settings



Settings for: - local connections -connection to a COM port or a LAN
- modem - remote control /telemonitoring
- confirmation - reading and entering DAT into the PBX

- Select communications device select COM, TCP/IP, UDP
- TCP/IP Setting IP addresses of PC where an XAPI server is running
- Selecting COM Selecting the COM ports
- COM Baud speed Setting the speed of communication with the PBX

- Select language Choosing the language you want to use

1.4. Help

- Contents help
- Search… for words in help
- Using help general instructions for using help in OS WIN
- About the application information on the SW version of the Program

2. Programming tables for the *2NÒ - OMEGA Lite* PBX

The following passages give a detailed description of all of the programming parameters in the *2NÒ - OMEGA Lite* PBX. The chapter is in the same position as the menu in the PBX configuration device's themed index.

For access to programme the PBX you need a password.
The PBX configuration instrument enables **two** types of users to change the PBX settings.

Password for a Supervisor user

The user with the **supervisor** authorisation has access to all of the PBX parameters and can also change the password for a user with **user** authorisation.

Password for an ordinary User

The **user** authorisation only allows certain PBX parameters to be changed when programming and the **User** password to be changed.
All parameters are available for reading.

The user can change the following parameters
- Authorisation numbers
- Emergency numbers
- Bank holidays
- The name of the VL
- Internal priority lines
- Authorisation without ARS
- Allowing services – all
- Divisions
- Classification into groups and divisions
- Calling to the entire division
- FLASH and the type of choices for VL
- Allowing direct choice for bundles
- Operating parameters for internal and external lines
- Buttons for system telephones
- External lines – name and type of choice
- Bundles for ring retrieval
- Assigning ringing tables into the ringing table for external lines
- Labelling lines
- Switchover times
- Global group data – dialling tone, softening tone, DISA tone
- Quick dialling – joint and personal

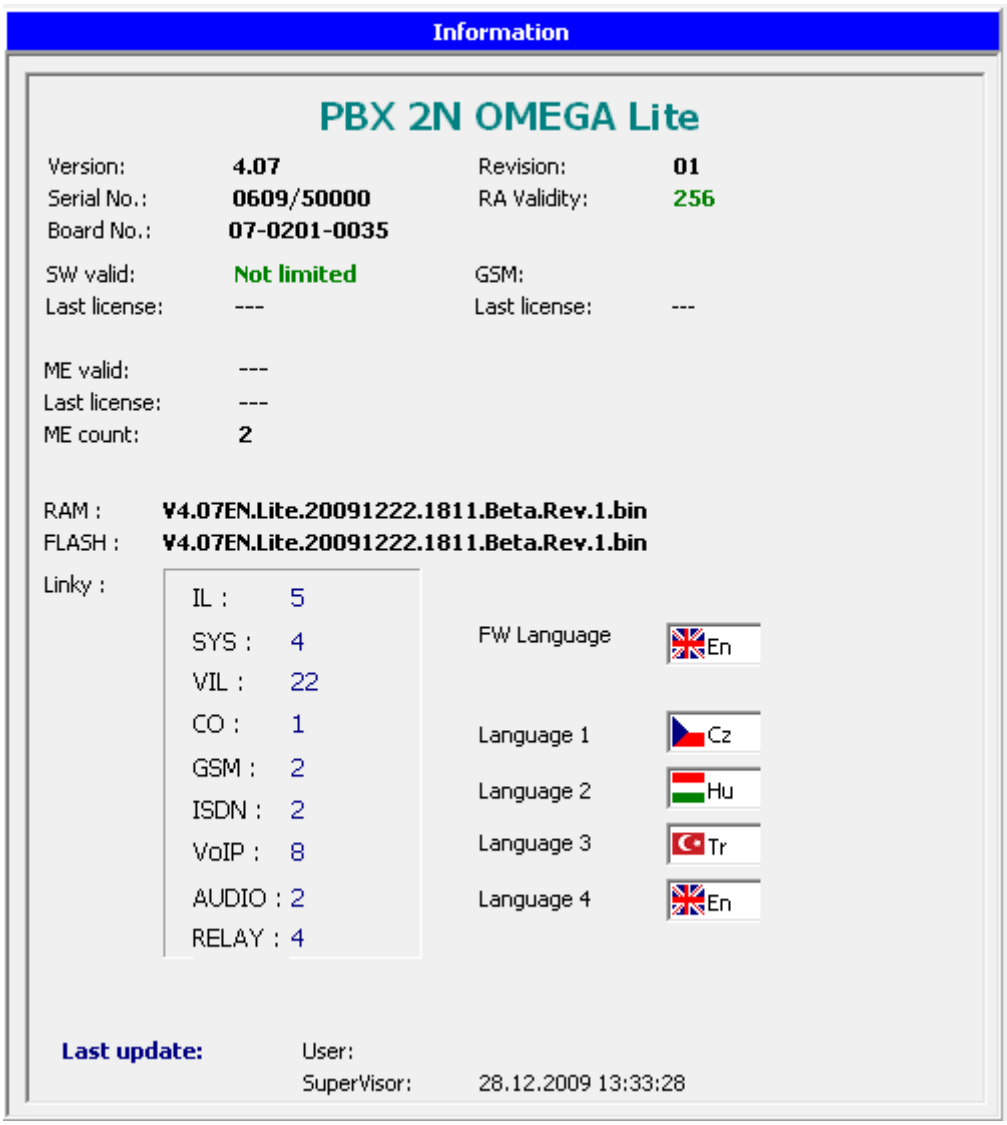
The standard password setting for both users is **1111**.

Without the password the PBX cannot be programmed. Please remember the password!

2.1. Global data

2.1.1. Information

This item informs you on the construction of the PBX that you are programming. Here you will find the number of the SW version in the PBX, the production number of the PBX to which you are connected, the time for which access to the PBX by remote control is allowed and a precise labelling of the PBX firmware saved in the RAM and FLASH memory. This labelling is in accord with the file name used to upgrade the firmware in the PBX.



Information about licences:

RA Validity - expiring period for remote control with integrated modem to PBX (according to main board type).

SW valid - SW validity expiring period (it is possible to make only emergency calls after expiring of this period).

ME valid - licence expiring period of Mobility Extension service.

ME count - maximum count of Mobility Extension service users. (default users count = 2, without validity limitation).

GSM validity – displays the time before licence expiry for call blocking from GSM modules (blocking of IMSI or first IMSI part). View the IMSI after the SIM card is read in the v LogSystem listing.

Last licence - shows the valid licence serial number. It is possible to download a licence with a serial number higher than the current one. Such a licence may be entered just once into the PBX.

To enter a licence, push the 'key' button on the main bar of the configuration tool. To create a licence, communicate the licence type, validity time and board number to the supplier.

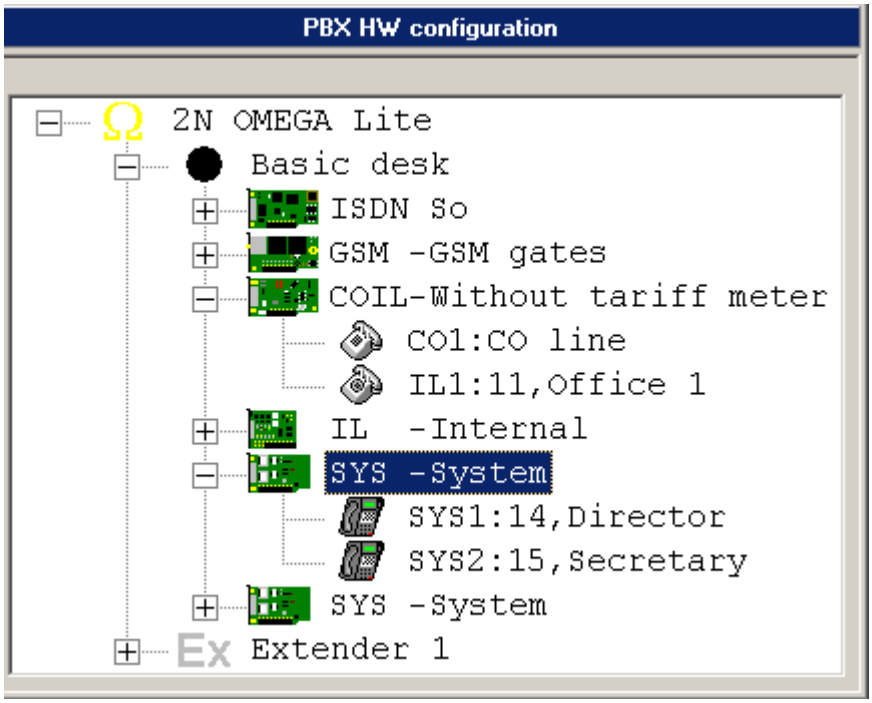
Lines - displays the count of available ports for each interface.

Languages - shows the system language to be used (for accounting and logsystem data listings and ME SMS messages as determined by the FW type: CZ = FW cz, sk, EN = FW en, de) and available user languages (for the key system menus and tags, extension names and personal quick dialling labels).

The **last changes** item provides information on the most recent data entered into the PBX or a file for each user type.

2.1.2. Hardware

Information on the physical construction of the PBX including the attachment of the individual types of modules at all positions in the PBX. Individual modules can be added or taken away thus altering the PBX's construction. However the simplest way is to read the actual configuration directly from the PBX along a series port or through the TCP/IP protocol.



2.1.3. Diagnostics for the PBX and localisation

If you use external and internal lines with call identification support, then fill in the part *Localisation:*

International – fill in the international calling code (usually 00)
National – fill in the country code in the national network (for the Czech Republic (CR) it is 00420)
Local – fill in the local transfer number in the ISDN network (usually 0, not used in the CR)

In the section *Installation* you assign who and where is installed in the PBX.

Languages

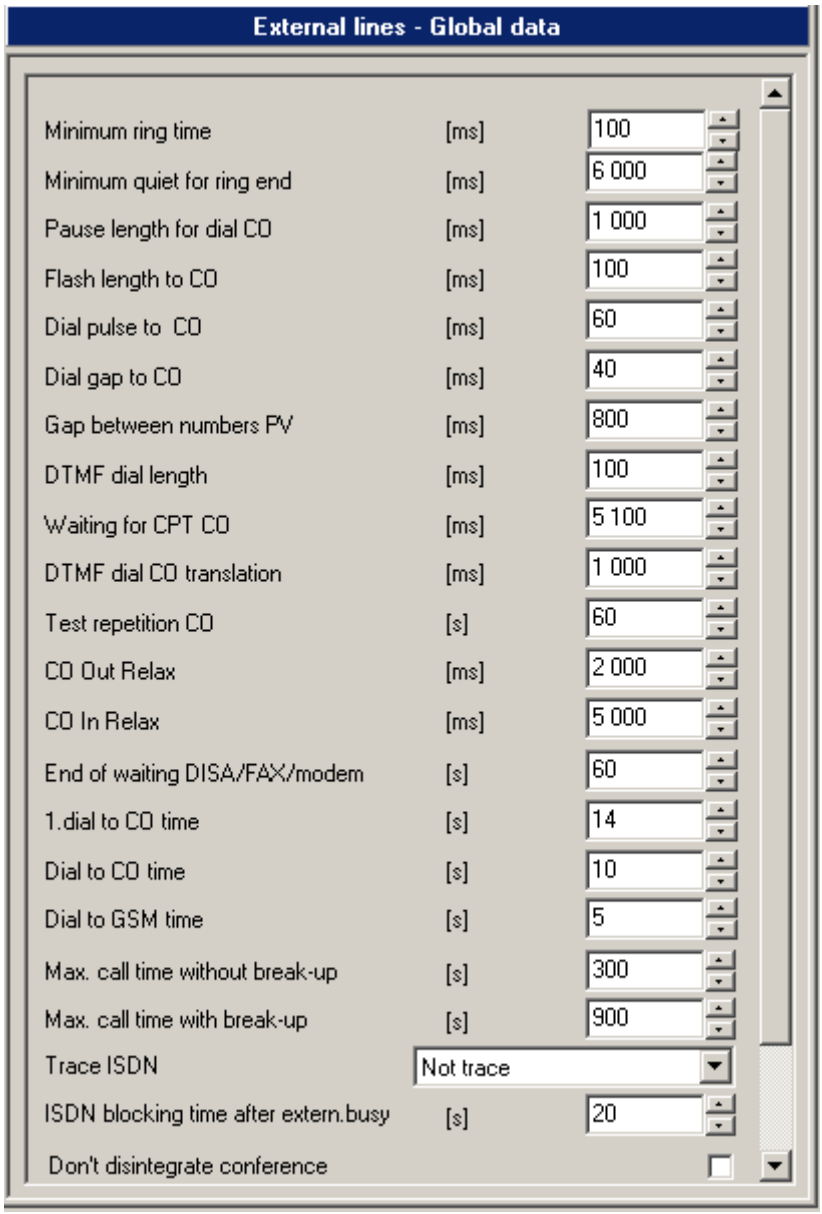
Select 1-4 user languages from the list of user languages (for key system phone menus and tags, extension names and personal quick dialling labels).

2.1.4. External lines

- **Minimum ring time.** Determines how long the minimum continuous ringing from an external line must be detected for the external line to be announced as ringing.
- **Minimum quiet for ring end.** It is the pause in the ringing from an external line after which the ringing from an external line is considered finished.
- **Pause length for dial CO.** Enables a pause to be entered between the numbers dialled. Used in complementary numbers after quick dialling and after detecting routes.

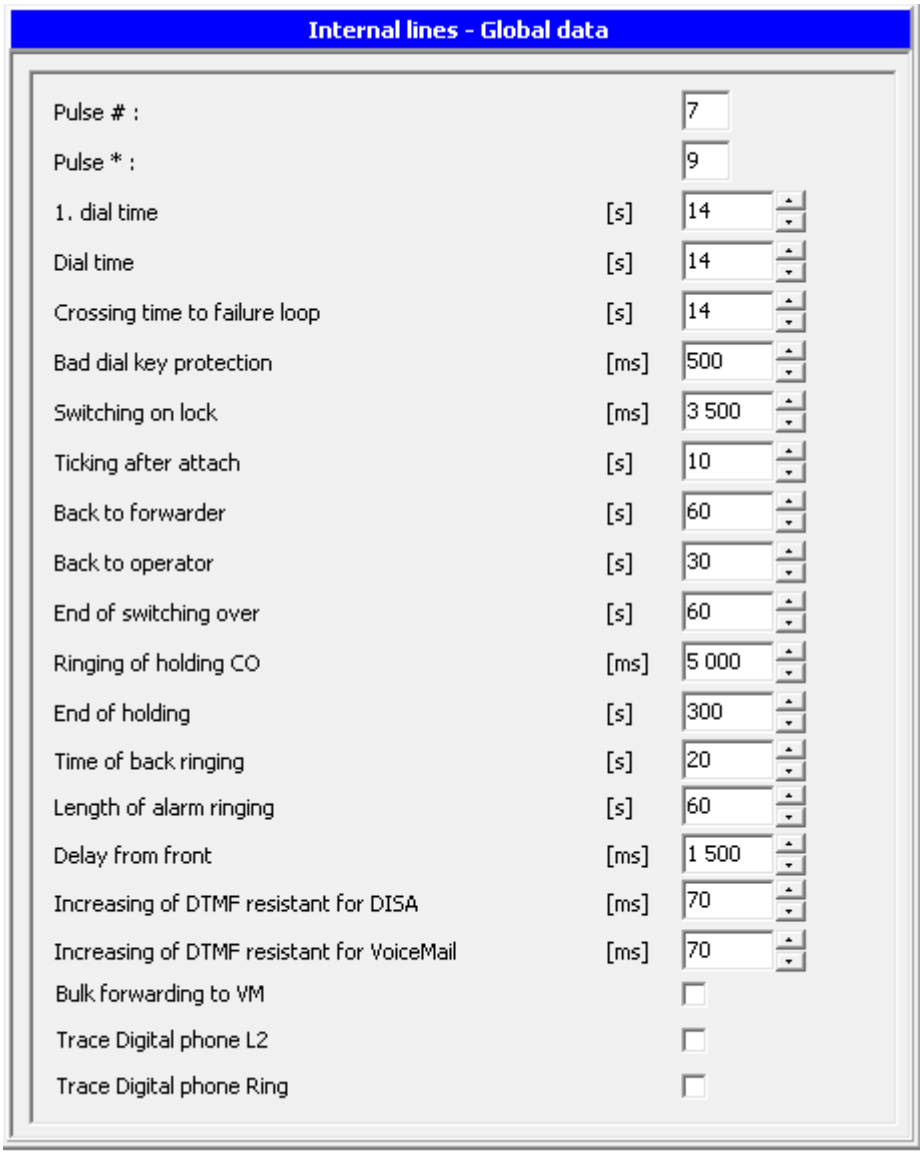
- **FLASH length to CO.** Enter the length of the FLASH signal generated by the PBX to the external port.
- **Dial pulse to CO.** Enter the length of the dial pulse's impulse to the external port.
- **Dial gap to CO.** Serves to set the gap length for pulse dialling to an external port.
- **Gap between numbers PV.** Serves to assign the gap length for pulse dialling sent to an external pulse.
- **DTMF dial length.** Serves to assign the length of broadcasting the dual tone multi-frequency (DTMF) dialling to an external port. This value also determines the DTMF dialling pause.
- **Waiting for a CPT CO.** Determines the maximum period the PBX spends trying to detect the external dialling tone used to determine the working of the external port.
- **DTMF dial CO translation.** The dialling gap for an internal tonal line to an external tonal line after which the dialling to the external line is only heard.
- **Test Repetition CO.** How much a faulty external line test is repeated.
- **CO OUT Relax.** The minimum time for suspending an external line after an outgoing call.
- **CO In Relax.** The minimum time for suspending an external line after an incoming call.
- **End of waiting on a DISA/FAX/modem.** The time an incoming external call waits after detection by the CNG (fax signal) or after direct inward calling on a free or engaged subscriber before hanging up on the external line or carrying on with another row in the call table.
- **1st dial to CO time.** The maximum time for delay when dialling an external line. If there is no dialling during this time the external number is hung up on.
- **Dial to CO time.** The maximum time for delaying in dialling an external line. It is renewed after every new number dialled by the internal user and, after it has expired; the device switches from dialling to calling. It is permissible for the expiration of this time to be marked by a short peep. Any other calling after this time has elapsed is considered to be a call service.
- **Dial to GSM time.** The maximum time for dialling another call to a GSM line. It is renewed after every new number dialled by the internal user and, after it has expired; the device switches from dialling to calling. It is permissible for the expiration of this time to be marked by a short peep. Any other calling after this time has elapsed is considered to be a call service
- **Max call time without break-up.** The maximum time for a call of two external lines that have connected break up signalling**.**
- **Max call time with break-up.** The maximum time for a call of two external lines that do not have connected break up signalling.
- **Trace ISDN.** The selection allows the type of diagnostics for the ISDN modules to be set after restarting the PBX.
- **ISDN blocking time after extern. busy.** The time for which the ISDN line is not accessible after a previous occupancy by external equipment (e.g. by an ISDN modem).

- **Don't disintegrate conference.** Allowing no collapse for conferencing between an internal line and at least two external lines after the internal line has hung up.

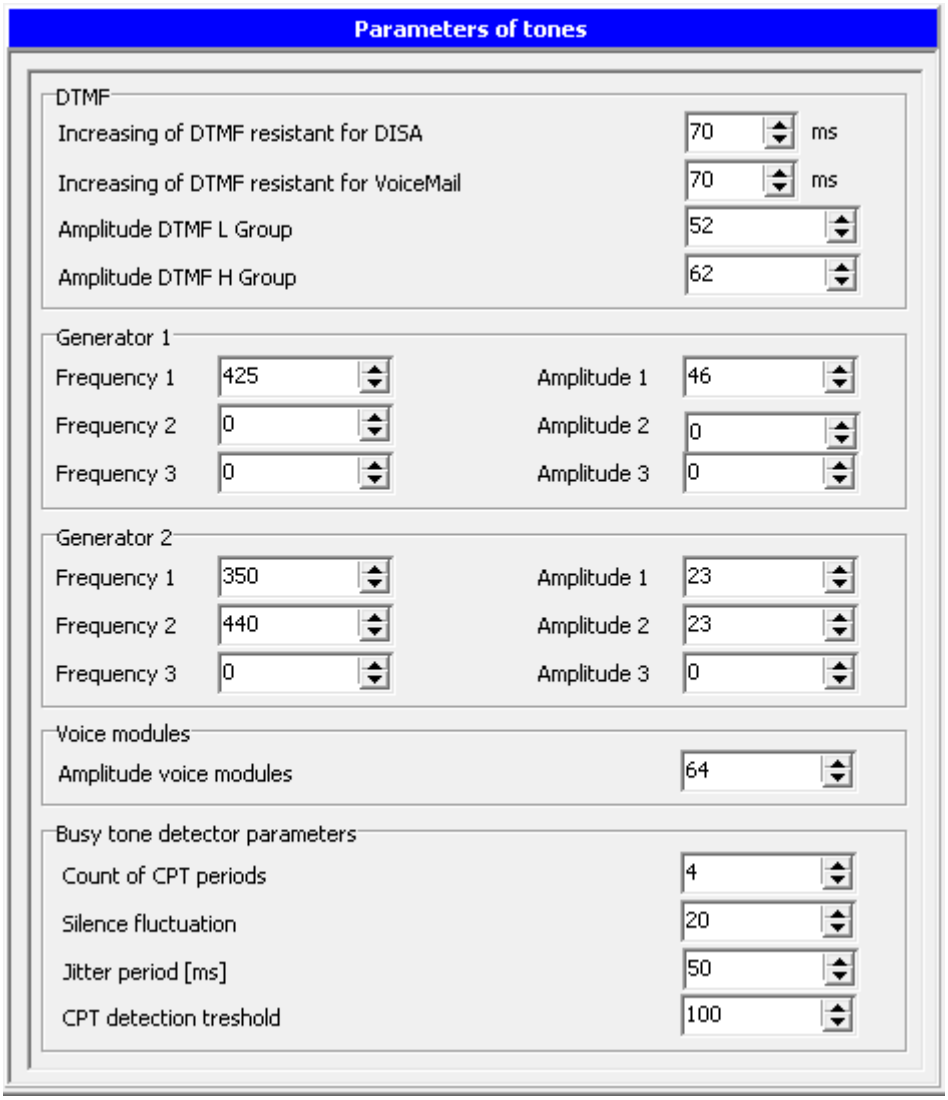

External lines - Global data
Minimum ring time [ms] 100
Minimum quiet for ring end [ms] 6 000
Pause length for dial CO [ms] 1 000
Flash length to CO [ms] 100
Dial pulse to CO [ms] 60
Dial gap to CO [ms] 40
Gap between numbers PV [ms] 800
DTMF dial length [ms] 100
Waiting for CPT CO [ms] 5 100
DTMF dial CO translation [ms] 1 000
Test repetition CO [s] 60
CO Out Relax [ms] 2 000
CO In Relax [ms] 5 000
End of waiting DISA/FAX/modem [s] 60
1.dial to CO time [s] 14
Dial to CO time [s] 10
Dial to GSM time [s] 5
Max. call time without break-up [s] 300
Max. call time with break-up [s] 900
Trace ISDN Not trace
ISDN blocking time after extern.busy [s] 20
Don't disintegrate conference

2.1.5. Internal lines

- **Pulse #.** The digit replacing the tone hash on pulse telephones.
- **Pulse *.** The digit replacing the tone asterisk on pulse telephones.
- **Time for 1st dialling.** Defines the maximum time waiting for the first number to be dialled after phone is picked up. The line gets the engaged tone after this time period has elapsed.
- **Time for further dialling.** Defines the maximum time spent waiting for the next number to be dialled from an internal line. After it has lapsed the engaged tone is heard and the call is cancelled.
- **Crossing Time to failure loop.** Defines the time period before the engaged tone is sent following unsuccessful dialling. After the line goes to a fault loop.
- **Bad dial key protection.** The time period for which the next digit dialled is ignored after dialling the entire number. It serves to prevent inadvertent call back after unsuccessful dialling.
- **Switching on lock.** The time the intercom lock is fastened.
- **Regime supervisor is valid.** The time period for which the supervisor services can be called upon after logging in to the Supervisor service
- **Ticking after attach.** The time for which a ticking can be heard on the call after connecting. After this period the service cannot be disconnected from the call.
- **Back to forwarder.** The time the PBX tries to get a connection after reconnecting with a hung up line. After this time it carries on in getting a connection but it also starts ringing back to the switcher.
- **Back to operator.** If the target user does not pick up the phone in this period the ringing goes back to the operator.
- **End of switching over**. After this period the PBX considers the call as dead. The caller is told this by a voice telling them that all lines are unavailable and the call is cancelled.
- **Ringing of holding CO.** The time between being notified by tinkling on an internal line that an external call is on hold.
- **End of holding.** The time limit for a temporarily held external line to be hung up.
- **Time of back ringing.** The time call back lasts for.
- **Length of alarm ringing.** The time the alarm will ring after activating it on an internal line.
- **Delay from front.** The time when a new incoming call is not accepted after having been hung up.
- **Increasing of DTMF resistant for DISA.** Min. length of the DTMF sign that is accepted in the DISA direct inward calling.
- **Increasing of DTMF resistant for VMail.** Min. length of the DTMF sign that is accepted in VoiceMail.
- **Trace key system phone L2** – enable writing of key system phone layer 2 events into LogSystem listings. To read the data push the ‘dollar’ button on the main bar of the configuration tool while reading accounting data, for example.
- **Trace key system phone ringing** – enable writing of key system phone ringing into LogSystem listings. To read the data push the ‘dollar’ button on the main bar of the configuration tool while reading accounting data, for example.


Internal lines - Global data
Pulse # : 7
Pulse * : 9
1. dial time [s] 14
Dial time [s] 14
Crossing time to failure loop [s] 14
Bad dial key protection [ms] 500
Switching on lock [ms] 3 500
Ticking after attach [s] 10
Back to forwarder [s] 60
Back to operator [s] 30
End of switching over [s] 60
Ringing of holding CO [ms] 5 000
End of holding [s] 300
Time of back ringing [s] 20
Length of alarm ringing [s] 60
Delay from front [ms] 1 500
Increasing of DTMF resistant for DISA [ms] 70
Increasing of DTMF resistant for VoiceMail [ms] 70
Bulk forwarding to VM
Trace Digital phone L2
Trace Digital phone Ring

2.1.6. Tone parameters



DTMF
Increasing of DTMF resistant for DISA. Min. length of the DTMF sign that is accepted in the DISA direct inward calling.
Increasing of DTMF resistant for VMail. Min. length of the DTMF sign that is accepted in VoiceMail.
Amplitude of the DTMF L Group: the size of the DTMF sign's amplitude (as a rule it is not necessary to change it).
Amplitude of the DTMF H Group: the size of the DTMF sign's amplitude (as a rule it is not necessary to change it)

The PBX has two tone generators which can be used as tone sources when setting *Tones*. The tone can be formed from up to three varying frequencies. When forming the tones it is necessary to uphold the rule on the overall size of the amplitude:
-1st Frequency = Amplitude to 62

-2nd Frequency = Amplitude to 31
-3rd Frequency = Amplitude to 21

Generator 1 (sine 1) **-**Default frequency 425 Hz.
Generator 2 (sine 2) **-**Example of forming a “British tone”.
Amplitude of the voice modules
The default value is set at 64 and it is not recommended to change it.

Busy tone detector parameters

Count of CPT periods -minimum count of tones (beeps) according to ITU-T E.180 before hang-up;
Silence fluctuation -intensity of the silence signal (0-65535).
Jitter period -tone duration tolerance against silence, starting from tone:silence = 1:1 [ms];
CPT detection threshold behind filter -minimum tone level (0-255) for the tone to be considered present.

2.1.7. Parameters for voicemail

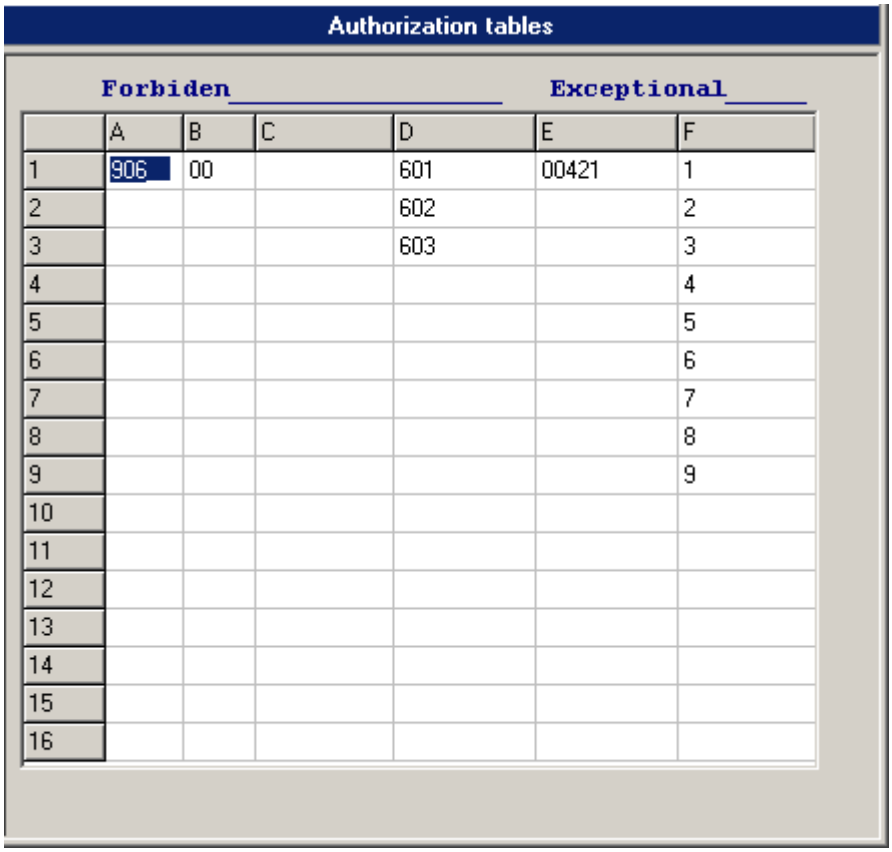
Voicemail a line (or department) to which the modem of a voice system is connected, Audio lines (Kerio VoiceMail) and the ITS voice system lines.
Identification of mailbox selection prefix before the identification sign (the default setting is a pause of 1s).
DTMF for call clear return from being put through (called subscriber is engaged)
Increasing of DTMF resistant for VMail. Min. length of the DTMF sign that is accepted in VoiceMail.

Use protocol ITS permission/prohibition for the voice mail ITS
Use protocol 602 permission/prohibition for the voice mail SW602

2.1.8. Authorisation numbers

Tables A, B, C, D, E, F for programming authorisations. Tables A, B, C, D are prohibitive and conversely tables E and F are for exemptions and have greater priority. Every table has 16 rows. Local numbers up to 5 digits can be entered into tables A and B, the rest take ten digits.

Apart from the numbers it is also possible to add the sign “?”, which presents all numbers (0-9).



	A	B	C	D	E	F
1	906	00		601	00421	1
2				602		2
3				603		3
4						4
5						5
6						6
7						7
8						8
9						9
10						
11						
12						
13						
14						
15						
16						

2.1.9. Emergency numbers

A table listing the emergency numbers that the lines can call without authorisation. 15 numbers are available with a maximum of 16 digits.

2.1.10. Passwords

2.1.10.1. The password for users classified as Supervisor

The table for entering the supervisor's password. This password is used, for instance, during the RESET services of the PBX etc.

The user with the **Supervisor** authorisation has access to all parameters in the PBX and can also change passwords for users with the **User** authorisation**.**

2.1.10.2. The password for users classified as User

The user with the **User** authorisation has the possibility of changing only some of the PBX parameters when programming and can change the password for the user with the **User** authorisation.
All parameters are available for reading.

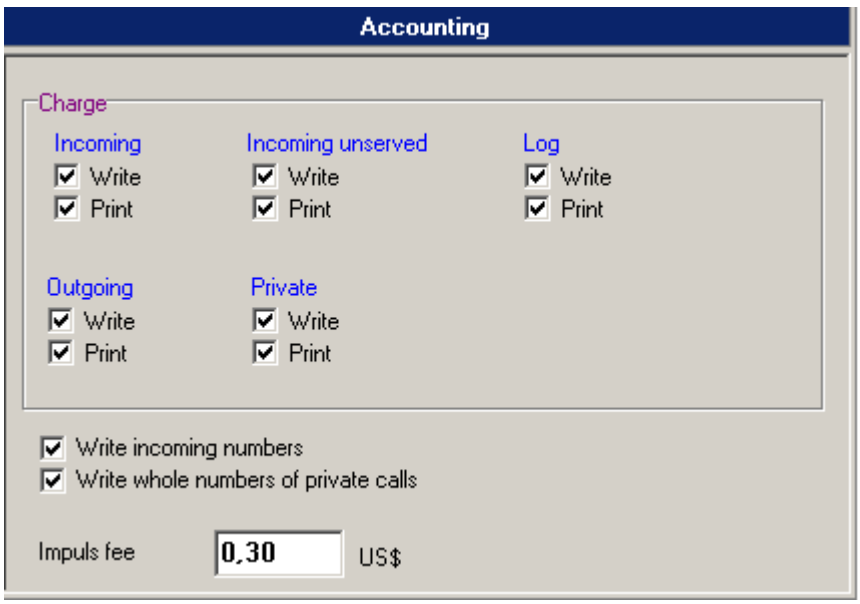
2.1.11. Bank holidays

A table in which up to 16 annual bank holidays can be entered. The PBX behaves as though it were a Sunday.

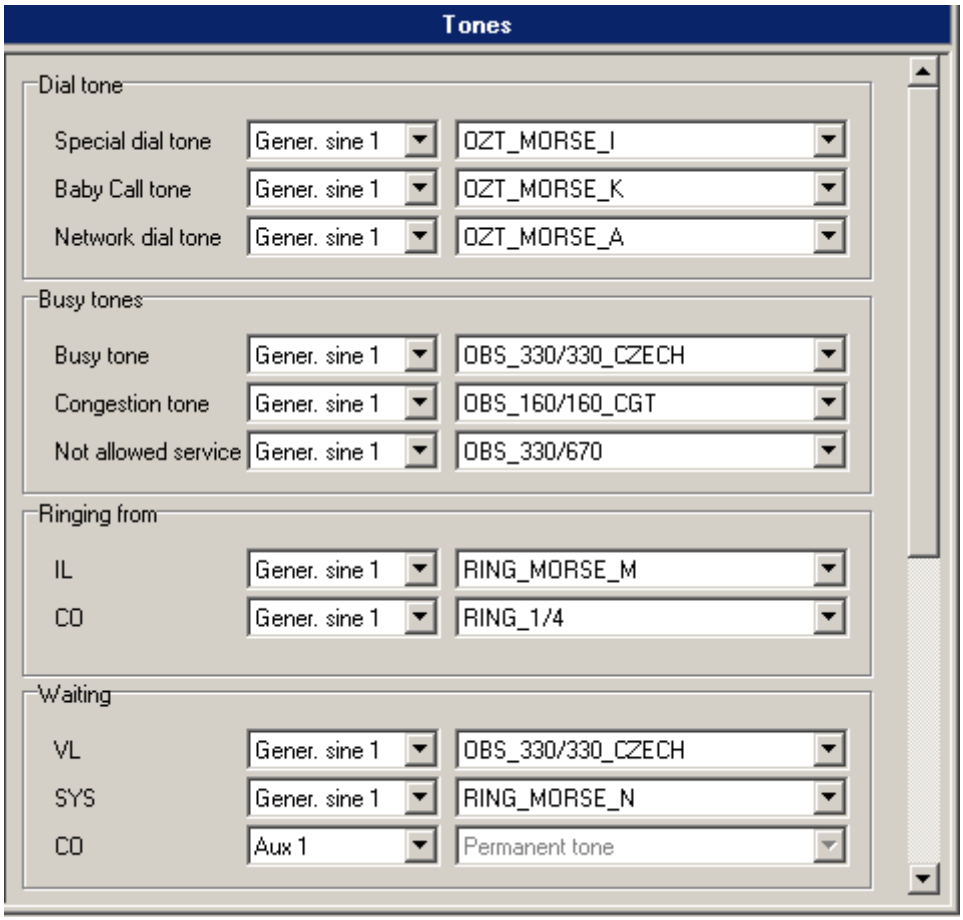
2.1.12. Charging

A form for entering the types of calls recorded in the call memory.

The entry **Record incoming numbers** allows a record of the incoming calls to be made in the accounting rows for ports so enabling this. The entry **Record the entire number of private calls** enables the numbers of private calls to be recorded.



2.1.13. Tones and ringing



In this screen you can change the type of tones and ringing in individual states. The screen is primarily meant for special applications.

It is necessary to select “Tone source” (left part of the table) and “Tone cadence” (right part of the table).

The types of tones (heard in the handset) and ringing (heard on the telephone):

Dialling tone

- Special held calls
- Baby call activate the Baby call/dispatcher regime
- Network dial tone simulated tone when using ARS, call back regime, internal GSM and ISDN in certain cases

Busy tone

- Busy the person called is engaged, the service cannot be provided
- Congestion blocked network (*from an external network* – the line is broken, network is blocked, wrong number no credit; *calling internal network* – telephone broken, full queue, cannot be redirected, reject)

- Not allowed service — unauthorised entry (entered an unauthorised service or line, lock, timeout selection, cannot be done)

Ringing from

- An internal line
- The intercom
- An external line
- Call back
- Return from redirection
- Alarm

Waiting tone

- VL — calling from an internal line
- SYS — calling from a system line
- CO — calling from an external line

Ticking

- When connecting
- End CO-CO — at the end of a call between two external lines there is a ticking before break up

Before entering this screen there is a warning about the possible negative effect operation may have on the PBX after an unprofessional change to the tones.

2.1.14. Voices

Here you can change the voice settings (system, user and music to make waiting more pleasant).

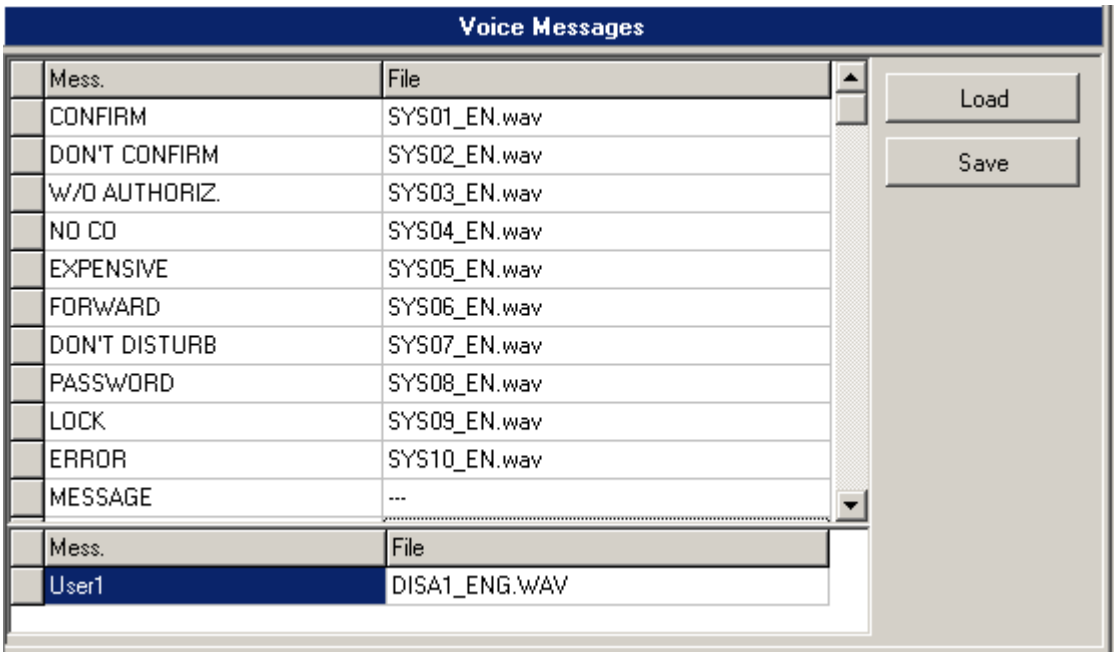
System voices

The PBX contains 11 systems voices. The voices notify the user about being redirected, accepting the service dialled, a more expensive call in the LCR (least cost routing) etc...
In this category there is also “music to make waiting more pleasant – MusicOnHold“.

It is not recommended to make changes to the system voices.

User voices

They are for notifying the caller in DISA.



After reading the DAT from the PBX this table will display a list of the voices contained in the PBX.

Pressing the key *“Load“* saves the voices in the “Wav“ directory, which are then automatically formed in the root directory of the installed PBX’s configuration tool.

If you are going to make changes in the voices settings it is necessary to first put the voice in this “Wav” directory and then set it here. **A voice** is a type of **wav** and must have the following parameters: **MONO - 8bit - 8kHz - unsigned**.

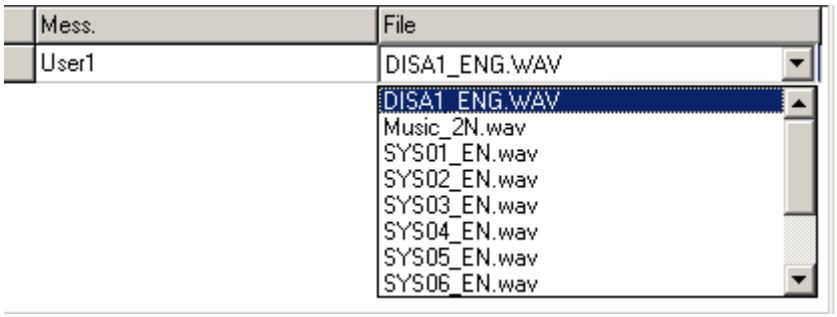
The overall capacity of the voices is ca. 3 minutes. The length of the system voices including the music is ca. 40 sec.

Mess
The name of the voice as it is herein presented in the PBX configuration tool (e.g. as a DISA parameter in the ringing table).

File
The name of the voice as it is saved in the PBX.

A change to the original voice

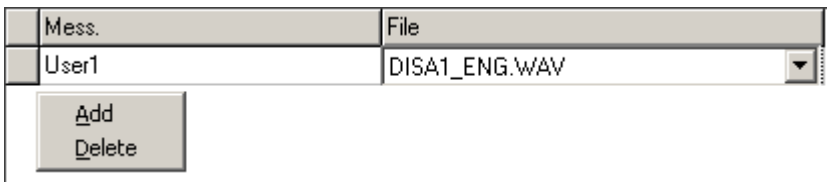
-press the key “Load”
-enter the voice created into the “Wav” directory formed.
-right click with the mouse on the voice you want to change.



-select the voice formed that is saved in the “Wav” directory.
-press the key *“Save”*

Adding a new voice

- press the key “Load”
-enter the voice created into the “Wav” directory formed.
-right click with the mouse in the space for user voices.



- select the command *“Add”*
- select the voice formed that is saved in the “Wav” directory.
- press the key *“Save”*

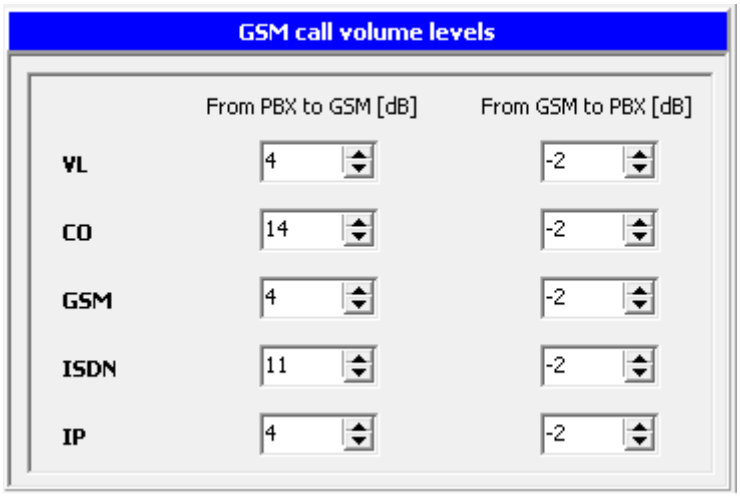
2.1.15. Modem

Changes can be made in the internal modem’s configuration (depending on the PBX model) for telemonitoring. The basic set up accepts silent running of the modem. The settings are changed by the AT command. The command is accepted after carrying out a RESET**.**

Permission for **Modem switch for telemonitoring** in the DISA detection is (according to the ringing table) selected (if it is sited) directly in the modem module.

2.1.16. GSM Call Volume Levels

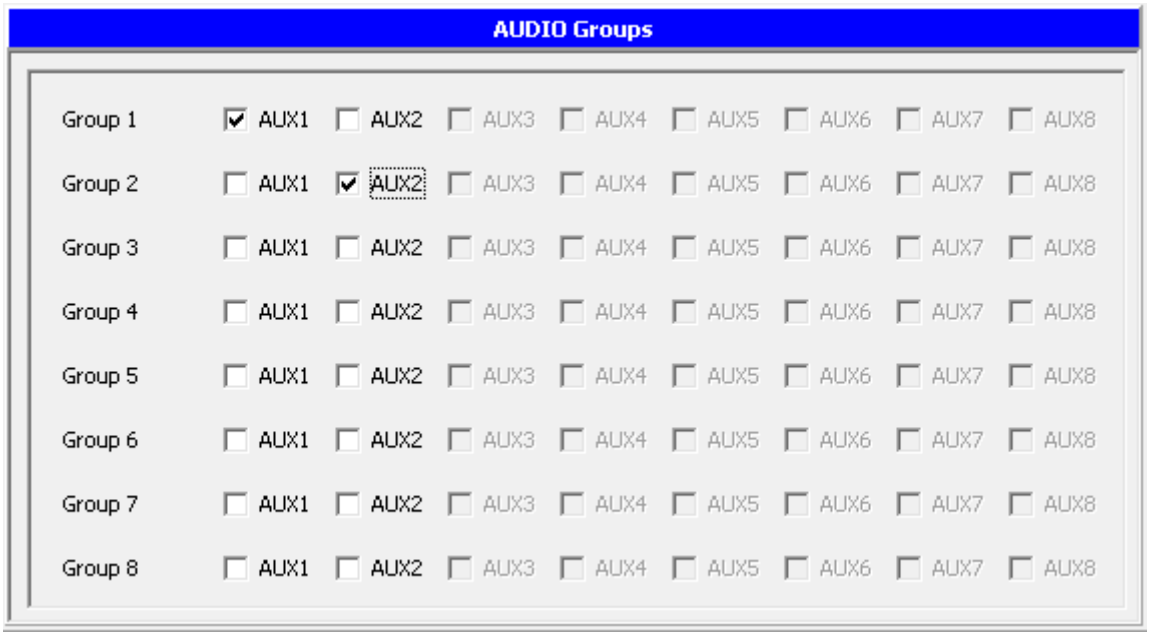
The table helps adjust the automatic call volume through a GSM module against other interfaces.



The values are in dB.
Take small steps while changing the settings.

2.1.17. AUDIO Groups

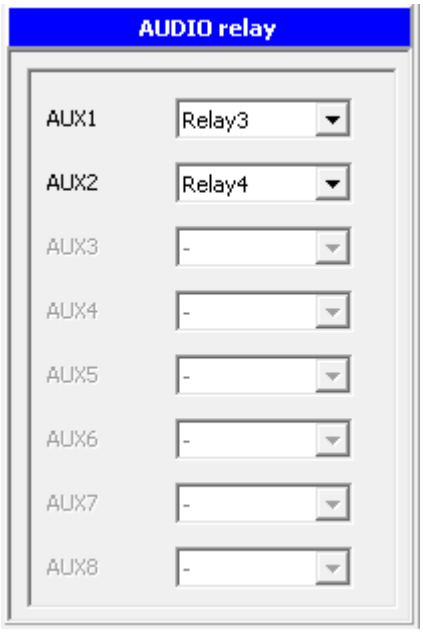
The AUDIO inputs/outputs include eight AUDIO groups to be assigned to the Broadcast/Tape recorder service in the *Groups/Switches, Broadcast, Tape recorder* menu.


AUDIO Groups
Group 1 AUX1 AUX2 AUX3 AUX4 AUX5 AUX6 AUX7 AUX8
Group 2 AUX1 AUX2 AUX3 AUX4 AUX5 AUX6 AUX7 AUX8
Group 3 AUX1 AUX2 AUX3 AUX4 AUX5 AUX6 AUX7 AUX8
Group 4 AUX1 AUX2 AUX3 AUX4 AUX5 AUX6 AUX7 AUX8
Group 5 AUX1 AUX2 AUX3 AUX4 AUX5 AUX6 AUX7 AUX8
Group 6 AUX1 AUX2 AUX3 AUX4 AUX5 AUX6 AUX7 AUX8
Group 7 AUX1 AUX2 AUX3 AUX4 AUX5 AUX6 AUX7 AUX8
Group 8 AUX1 AUX2 AUX3 AUX4 AUX5 AUX6 AUX7 AUX8

2.1.18. AUDIO Relay

Each AUDIO input/output can be assigned one relay. The relay closes when the given AUDIO channel is used for the Tape recorder/Broadcast service.

Typically, this option is mainly used for Broadcast with the relay activating the so-called Broadcast exchange.



2.2. Internal lines

2.2.1. Numbering

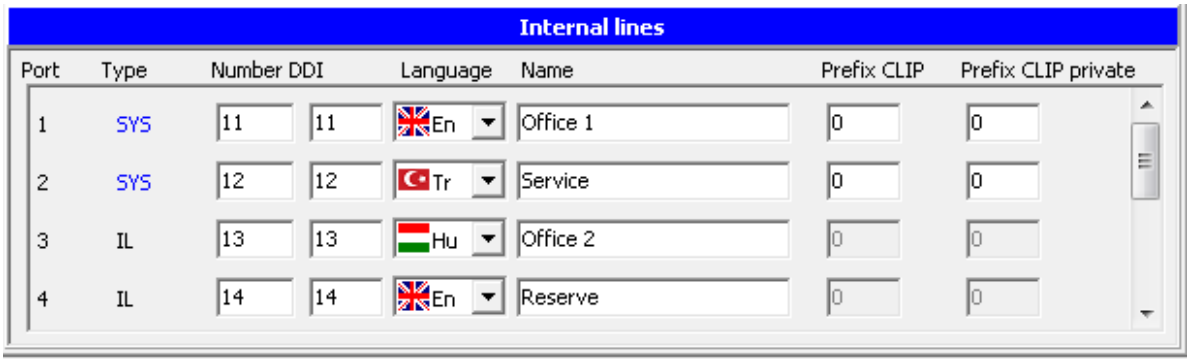
This item offers a list of internal subscribers to the PBX. Here the individual ports are assigned a subscriber number and a name. The name is allowed to up to 14 characters, the numbering is 4 digits at most. Pay attention that no collision with the numbering of services arises.

In the **DDI** column enter the DDI number used when phoning an internal line during DISA, DDI, if it is not set up otherwise in the ringing table.

Select the user language in the **Language** column to display the key system phone menus, line names and personal quick dialling labels. Select one of up to 4 languages as preset in the *Global data/Localisation* menu.

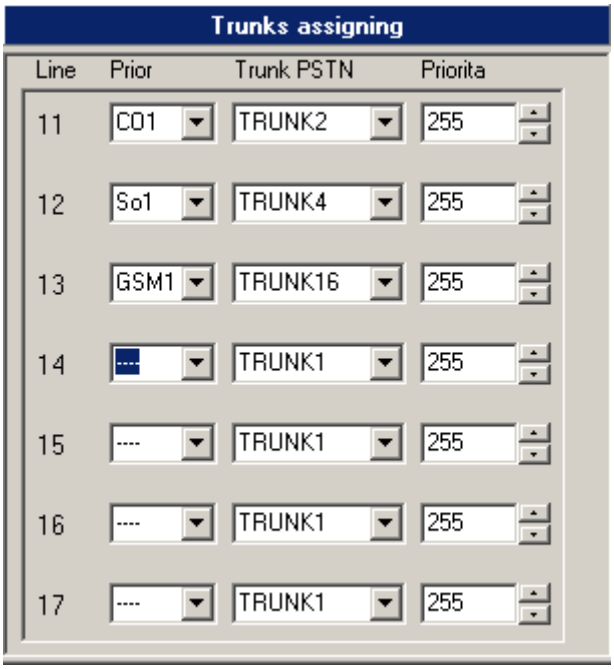
The **Prefix CLIP** column represents the transition sign to the VTS for internal lines that identify an incoming call and for system telephones with at least 05.12 firmware. It enables this equipment to accept call back directly from a list of rejected calls. It can contain up to 4 characters.

The column **Prefix CLIP private** represents the transition sign to a private network (an external line denoted as a private one) for internal lines that identify an incoming call and for system telephones with at least 05.12 firmware. It enables this equipment to accept call back directly from a list of rejected calls. It can contain up to 4 characters.



2.2.2. Access to external lines

2.2.2.1. Allocating external lines

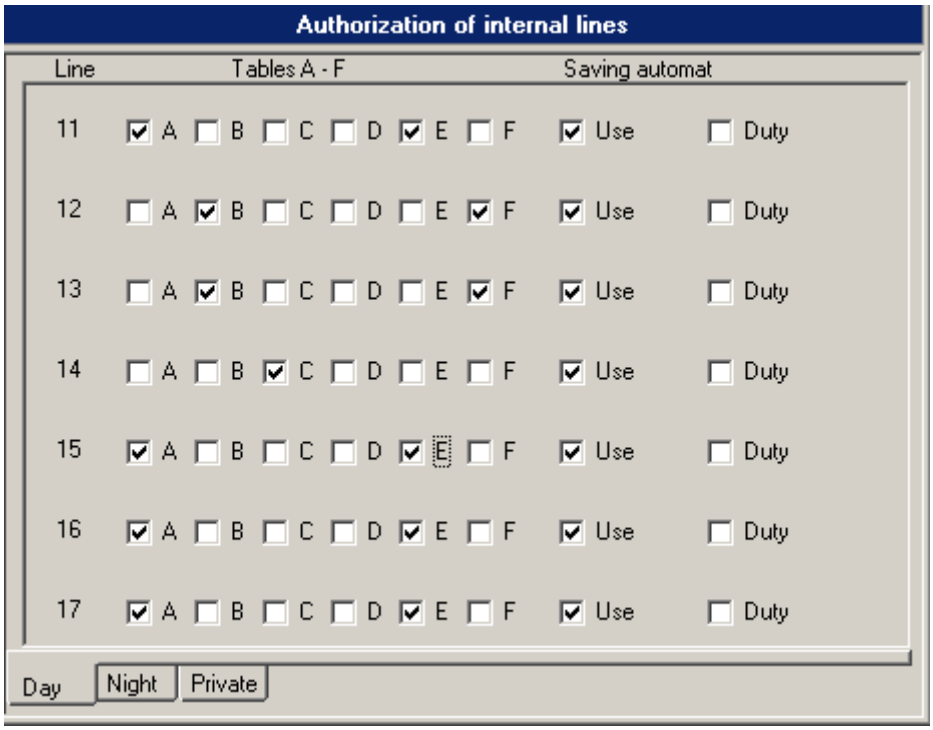


This table is for allocating a selection of bundles for external lines to internal lines (analogue and system) subscribers. If an internal subscriber chooses the service **Access to an external line** and the LCR is not activated or they are returning from it because a different, cheaper route was not found, then for calling to the VTS the **priority** external line is first used and if the first choice is not available then the first free external line selected from the bundle. The priority line can even be from a disallowed bundle.

If, for instance, we imagine 4 varying privileged groups of subscribers from the standpoint of their choices for a free external line (ignoring the LCR). The first circuit will be 2 internal lines to the flat of the building administrator, the second circuit to the director with an assistant and a company fax, the third circuit will be 4 traders and the fourth circuit the other employees with the lowest set selection for an external line. In order to meet these requirements we split up the external lines (except external lines allocated to GSM phones) into, for instance, 4 bundles. Bundle 1 has 1 external line – for the building administrator's flat. External lines 2

and 3 – the directorate – are assigned to bundle 2. The external lines 4 and 5 are in bundle 3 and the last external lines, 6, 7 and 8 are assigned to bundle 7. Afterwards for starting up an external line we can assign the internal links from the administrator's flat just to bundle 1. Like this no one else can get on this external link. For the director, assistant and fax we assign bundles 2 and 7 for starting up to the VTS. Like this two calls can be made simultaneously from the directorate on the private lines and other calls (faxing, other external conference calls , …) choose one of the free external lines from the last bundle, 7. Similarly the traders' lines can be chosen from bundles 3 and 7 for external calls. The other employees can only use the last bundle, 7. In this case bundle 7 not only serves the group "other employees" but also as a reserve for the traders and the directorate with a fax. This selection can be enriched by setting the priorities for the internal and external lines. For the priorities the valid principle is that an internal subscriber must have the same or higher priority than an external line so that it can be assigned to him.

2.2.2.2. Authorisations


Authorization of internal lines
Line
Tables A - F
Saving automat
11 A B C D E F Use Duty
12 A B C D E F Use Duty
13 A B C D E F Use Duty
14 A B C D E F Use Duty
15 A B C D E F Use Duty
16 A B C D E F Use Duty
17 A B C D E F Use Duty
Day
Night
Private

This table sets the authorisations for calling to a public telephone network in particular each internal line for the **DAY** regime, the **NIGHT** regime and **private** calls in an arbitrary regime. Here it is also determined whether a line will obligatorily use the LRC or not and what level of savings for route selection.

If tables A – F are checked, then it means that the given internal subscriber controls the selection to an external network on the agreement of the number called with the values shown in tables A – F. If the number called is in agreement with the number in the prohibition table, the call is ended. If the caller has the prohibition table checked, then they can only call the numbers entered into the table. If the

combination and permission tables are combined then the permission table always has preference.

If the choice **Use LCR** is not checked, then after selecting the start up service to an external line the direct free external line is allocated without using alternate routes in the LCR. Without this choice the PBX behaves as standard. If this choice is checked so that after selecting the service start up to an external line no port picks it up, the caller receives a simulated external dialling tone and carries on. After every number called the selection is compared with the numbers in the **Dialling Analysis** table and at the same time the LCR level of the internal subscriber and the dialling analysis are compared. If a route is found then a call is made along it. If the route found is engaged or no alternative route is found at all or the level of the LCR is not in agreement then the call takes place along a standard external line with the announcement “Attention please, this call is more expensive”

Selecting the **Obligatory LCR**. If this box is checked it means the same as in the previous example, that after selecting the start up service to the VTS a simulated dialling tone is given and the number called is checked to see if it is not the same as a code in the **Dialling Analysis** table. If a number is dialled that is not in the **Route Analysis** then a connection is made through a standard external link. If agreement is found and the route is free, then the call is made along this route. In the event that the route is engaged, the caller is not allowed to make the call and receives the engaged tone or an announcement that the connection failed.

The selection **Mask for Dialling Analysis – Level of Saving**. Here up to 4 levels of saving. The individual rows in the **Dialling Analysis** also have 4 boxes for ticking the level of saving. So that the internal subscriber can get to the **Dialling Analysis**, they must have at least one box ticked for the level of saving in agreement with the row in the **Dialling Analysis.**

Note:

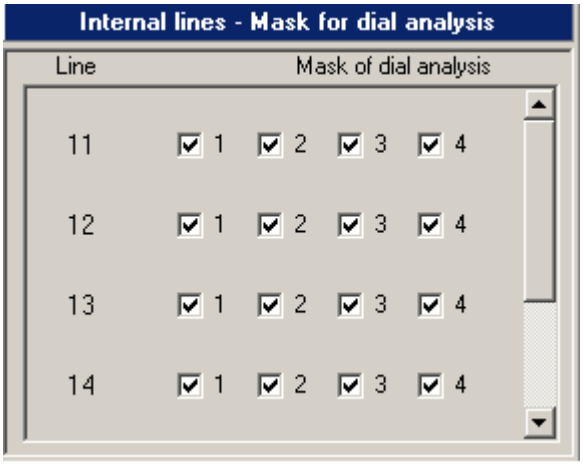
- *Setting the authorisations is independent of the LCR.*
- *The selection **Obligatory LCR** must be selected at the same time as **Use LCR**.*

Example 1: The subscriber is not allowed to call abroad. Then table B is ticked in which only one row is filled in with the value “00”. The other boxes are not ticked. Thus if they use “00” as the transfer sign to the international network then the call is ended.

Example 2: The subscriber is allowed restricted external calls, they are not allowed 906, they can only get to GMS Paegas through the GMS phoning module. Then they have Tables A and B ticked, in which only rows are filled in with “00” and “906” and they also have the boxes **Use LCR** and **Obligatory LCR** ticked. At the same time we assume that they have the first of the four savings levels ticked and the row **Analysis of Savings** is also ticked at the first of the four levels. After dialling 603 it is found that there is agreement with a route though GSM and if the route is free then the call is made. The route analysis is filled in so that in one row there is the code 603 followed, for instance, by the number for route 1. Thus the route number is programmed as to which bundle is used and the entries for adding and removing numbers are left free (see programming the LCR). If the GSM route

is engaged, then, because the subscriber has the LCR set as obligatory, they hear the engaged tone and will have to call later.

2.2.2.3. Mask for dialling analysis



This serves to set up one to four levels of saving for individual internal lines after the service *Dialling to the VTS*. Details in chapter *LCR/Level of Saving*.

2.2.3. Permitting services

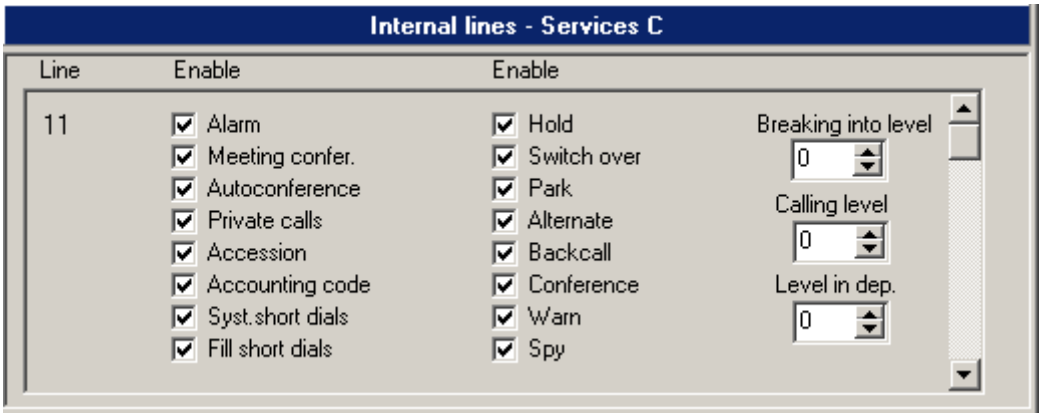
In this table it is possible to allow or disallow each internal line an individual service.

In group C it is possible to set three types of levels:

- connection level - for connecting to a call (default service = to call #0). A line that is at a lower or equal level can be connected to. The level has values from 0 – 15. The default level is 0, no one can call anybody.

- call level – for calling between internal lines. A line that is at a lower or equal level can be called. The level has values from 0 – 15. The default level is 0, everybody can call everyone.

- the level in the department – determines the ranking of lines to the departments. The lines to the departments are ranked from the highest levels to the lowest. This service is only tied in to automatic logging in to the department using the DSS key on the system phone. If the department is created by hand using the PBX configuration instrument then this service is not paid heed. The level has values from 0 – 15. The default level is 0.



The permission to receive CLIP on an analogue telephone is carried out in the marker activation.

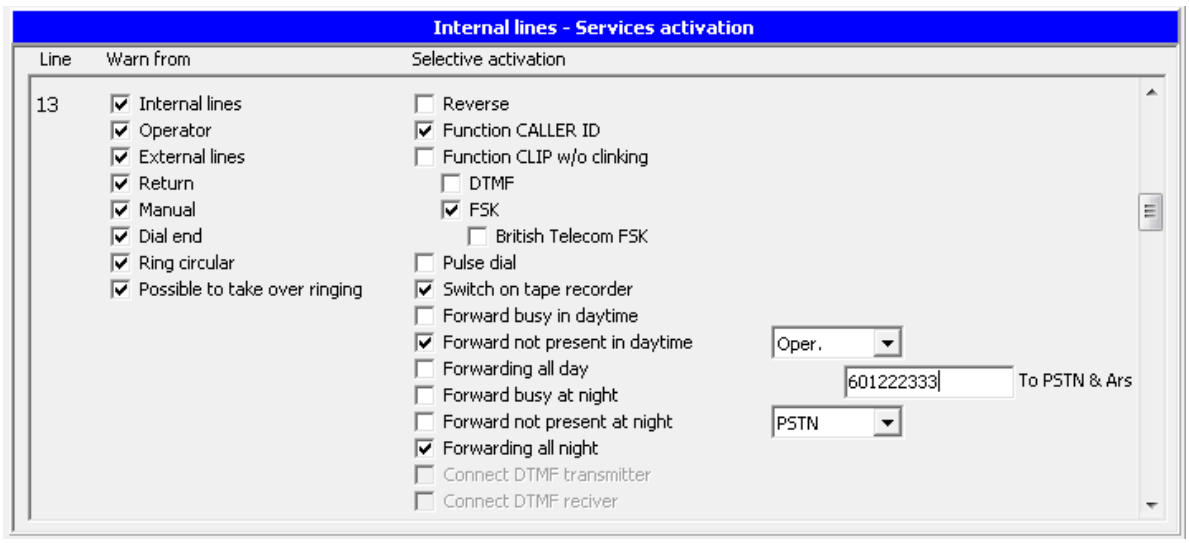
- CLIP function – by setting this CLIP will be displayed on analogue telephone (if the analogue telephone's connection allows this).

- **CLIP function without clinking** – in certain cases (wireless phones, MICROCOM, TOPCOM,..) to receive CLIP it is necessary to set the option of without ringing. This suppresses the first ringing and the accepted CLIP is displayed on the second ring.

- DTMF – setting this means the identification of the calling CLIP is sent using DTMF.

- FSK - setting this means the identification of the calling CLIP is sent using FSK (the majority of telephones with the CLIP display option support it)

If the telephone is connected with the BT CLIP standard then it is necessary to allow the **BT FSK** setting.



-Switch on tape recorder - enable this item to connect all outgoing/incoming calls to the appropriate AUX output as preset in the *Global data/AUX groups* and *Groups/Switches, Broadcast, Tape recorder* menus.

-Call forwarding xxx - select one of the administrator forwarding settings (always, busy, no answer for Day/Night modes). For call forwarding to the public switched telephone network, enter the required number into the *To PSTN and ARS* line.

***Attention**: On lines defined as fax, modem or intercom call notifications and peeping, peeping after the call to an external network has ended and round calls is not automatically banned. It is necessary to set it up for each individual line separately.*

2.2.4. Departments

These tables are used during the **Call** Department service. In all there are 8 tables with 56 fields available.

2.2.5. Allocating to a group and a department

The table serves to allocate individual internal subscribers to one of two groups for the purpose of switching from the DAY and NIGHT regimes. One subscriber cannot be in more than one group. At the same time this is where individual internal subscribers are allocated to one of eight departments for sorting ringing. One subscriber can be in more than one department simultaneously.

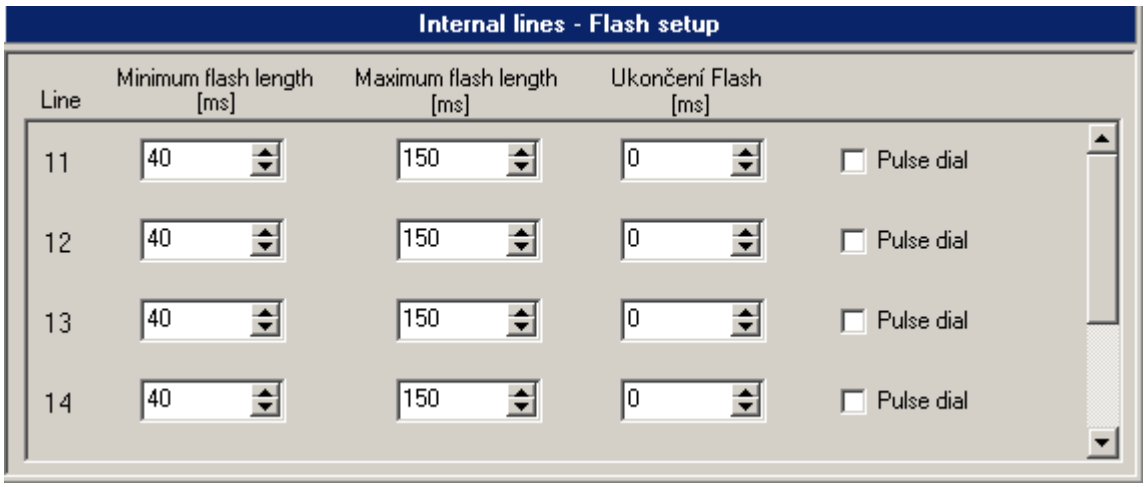
2.2.6. Calling to a department

If an internal line is using the call a department service, then the call either only starts on the first free internal line or at the entire department depending on which group the line belongs to and whether the given department has checked the parameter **Call to the entire department.**

2.2.7. Global data

See *Global data / internal lines*

2.2.8. FLASH settings and the type of dialling



The table is used for setting the minimum and maximum length of the FLASH on the analogue internal lines. FLASH detection is carried out by windows. A FLASH signal is only considered to be a disruption to the current by the subscriber loop that corresponds to the time set in the minimum and maximum length of the FLASH. By right clicking on the box with FLASH values it is possible to set the same value in the remaining boxes in the column.

The field “End Flash” is used when it is necessary to detect a break in the connection to an analogue CO-VL division. The connected analogue CO line must

be modified for detecting a break in the connection using a disruption in the current loop.

If the box **Pulse** dialling is not ticked then it only accepts tone dialling.

2.2.9. Allow direct dialling to bundles

This table is used to disable the service of direct dialling to a bundle. The internal subscriber is only allowed the service to the bundles that they have ticked.

2.2.10. Work and private MSN

Every internal line can be allocated to a different MSN for work and private calls. This means that during the regular settlement of costs for the operator's network costs can be divided up by the individual MSN.

2.2.11. Operating parameters

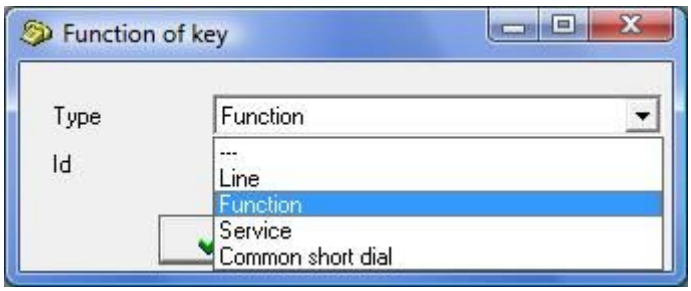
See *Operating data / Internal lines*

2.3. System lines

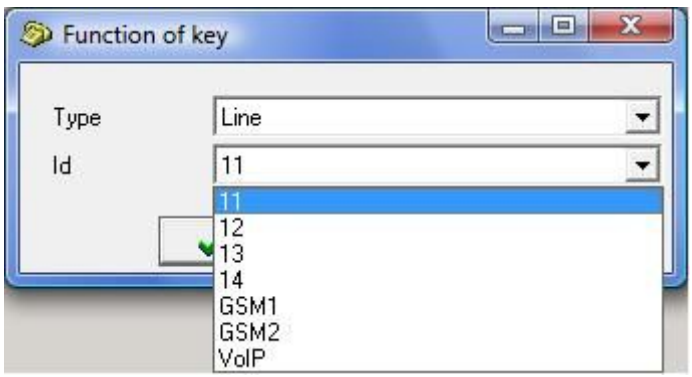
2.3.1. Buttons

In this MENU it is possible to set the programmable buttons for all of the connectable types of telephones. Two types of analogue system telephones (OMEGA FCI (no longer made), OMEGA LH) and a digital system telephone 2N StarPoint 500 (up to 5 types supplied) can be connected.

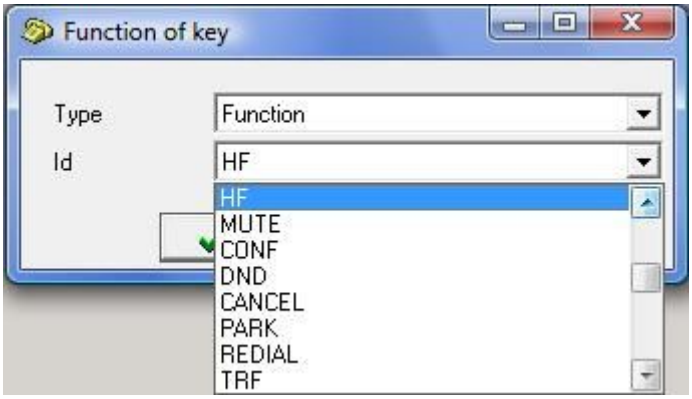
If you want to assign one of the functions (HandsFree, Transfer, Mute,..) to a programmable button, a direct call to an internal line or direct access to an external line then first left click the mouse on the programmable button that you want to set:



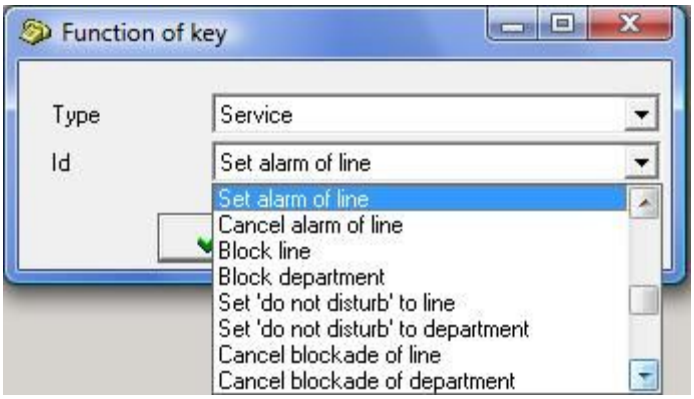
Select *Type*:
- Line (direct access to external and internal lines)



- Function (the basic function for controlling a system telephone – HF, MUTE, TRF, CANCEL,..)

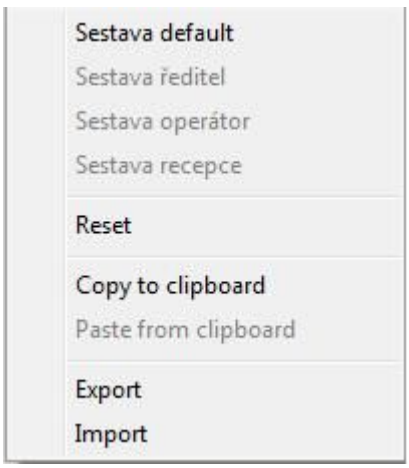


- Service (services – e.g. switching from DAY/NIGHT, logging in to a department, allowing Mobility Extension,..)



This is the procedure to set all of the programmable buttons as you wish.

If you right click with the mouse then you will see:



- Default setting, the programmable buttons' original setting (stable setting with regards to the individual types of setups).

- Manager setting, a specific setting for the programmable buttons if at least one extender is connected.
- Operator setting, a specific setting for the programmable buttons if at least one extender is connected.
- Reception setting, a specific setting for the programmable buttons if at least one extender is connected.

- Cancel, annuls the programmable buttons' settings.

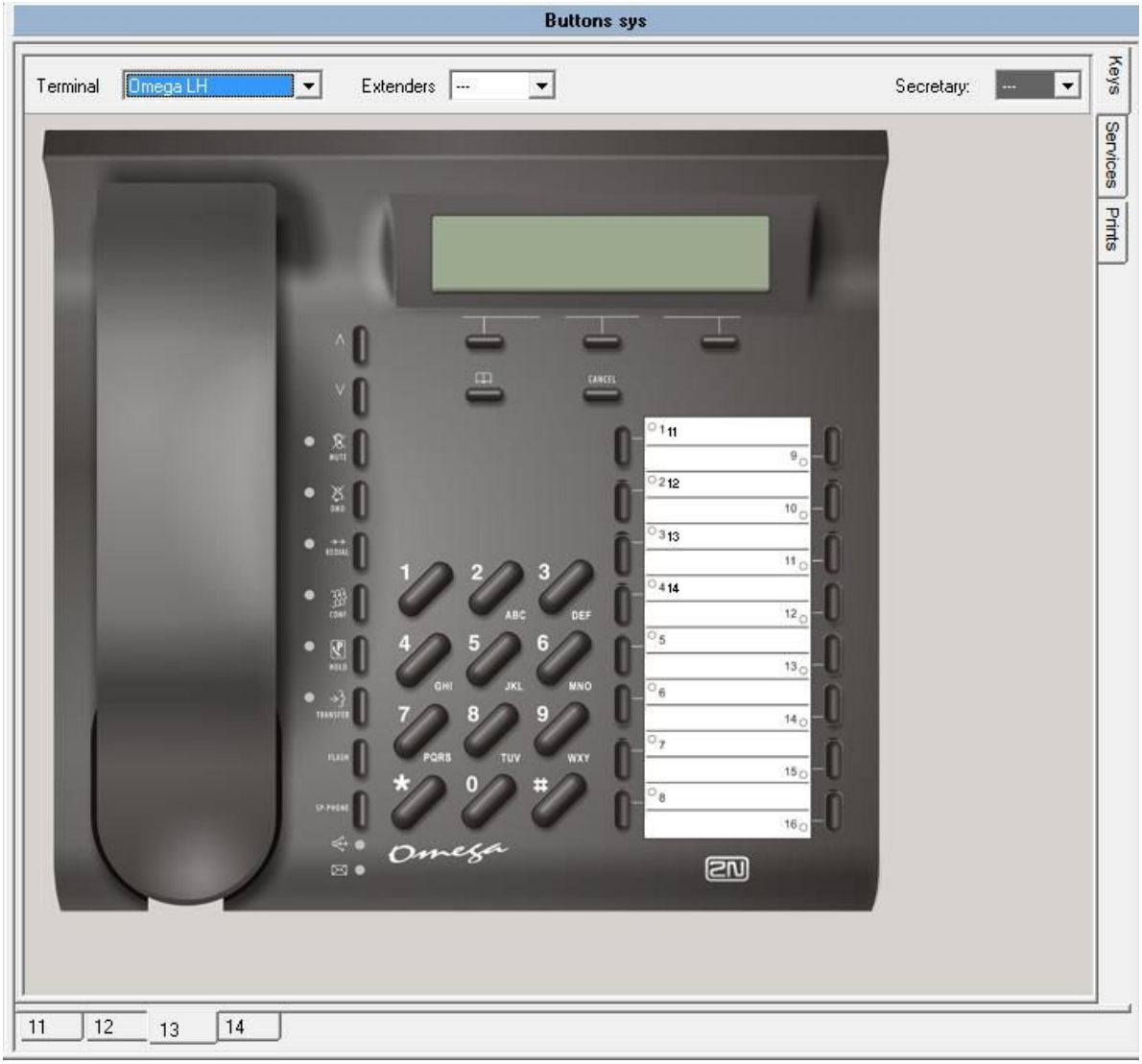
- To clipboard, copies the programmable buttons' settings.
- From the clipboard, enters programmable buttons settings that have been copied beforehand.

- Export, exports programmable buttons settings to a ".csv" file.
- Import, imports programmable buttons settings from a ".csv" file.

2.3.1.1. OMEGA FCI

Up to 20 programmable buttons can be set.

2.3.1.2. OMEGA LH



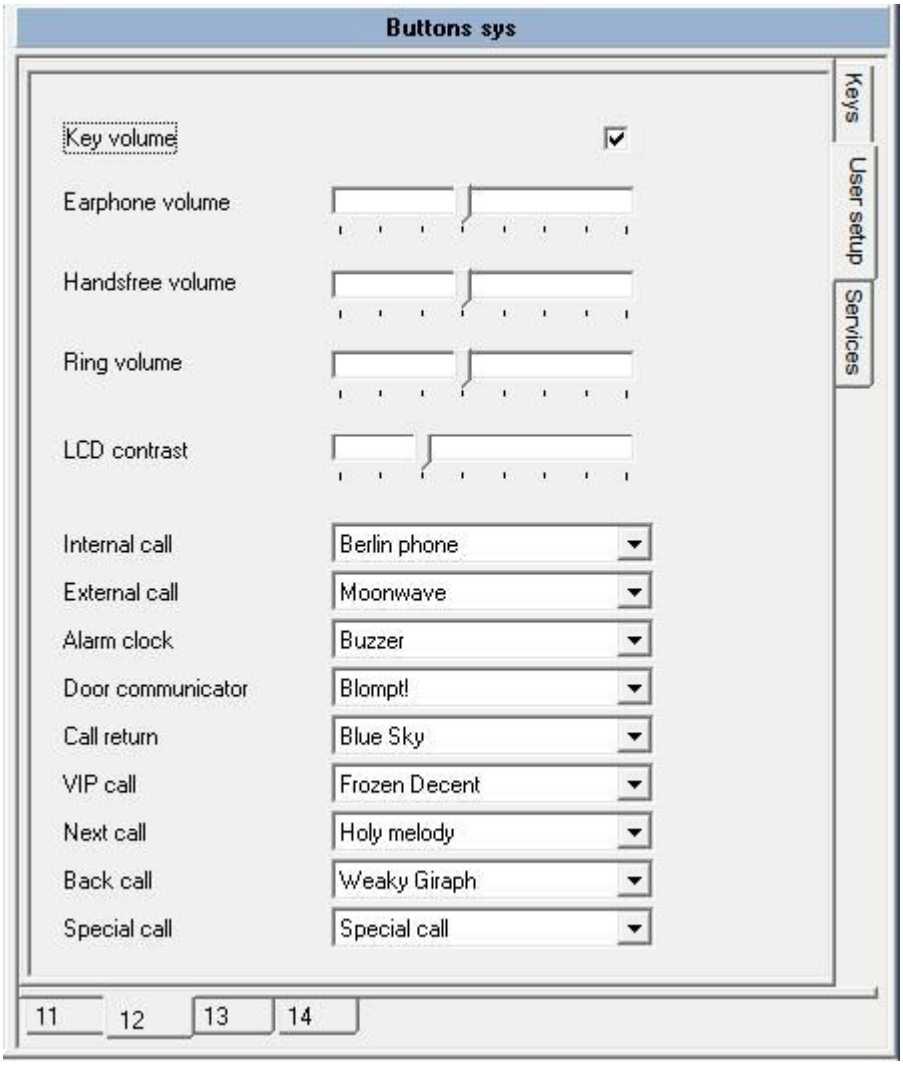
Can also be set with extenders (16 buttons = telephone + 2x20 button Extender), up to 56 programmable buttons.

2.3.1.3. 2N StarPoint 500

It can also be set with the connected extenders (19 buttons = Advance telephone type + 1x90 button Extender), up to 109 programmable buttons. Overall number of all programmable buttons can be up to 460 for the entire system.

For 2N StarPoint telephones it is possible to select from several ringtone melodies.

The melodies for individual types of calls can be assigned in the environment *PBX Configuration tool /system lines /buttons/user settings.*



Set:

- **Key loudness**, allows an acoustic signal when pressing system telephone buttons.
- **Earphone volume**, volume control for the earphone.
- **Handsfree volume**, volume control when using handsfree.
- **Ring volume**, the volume when ringing.
- **LCD contrast**, the intensity of the display contrast.

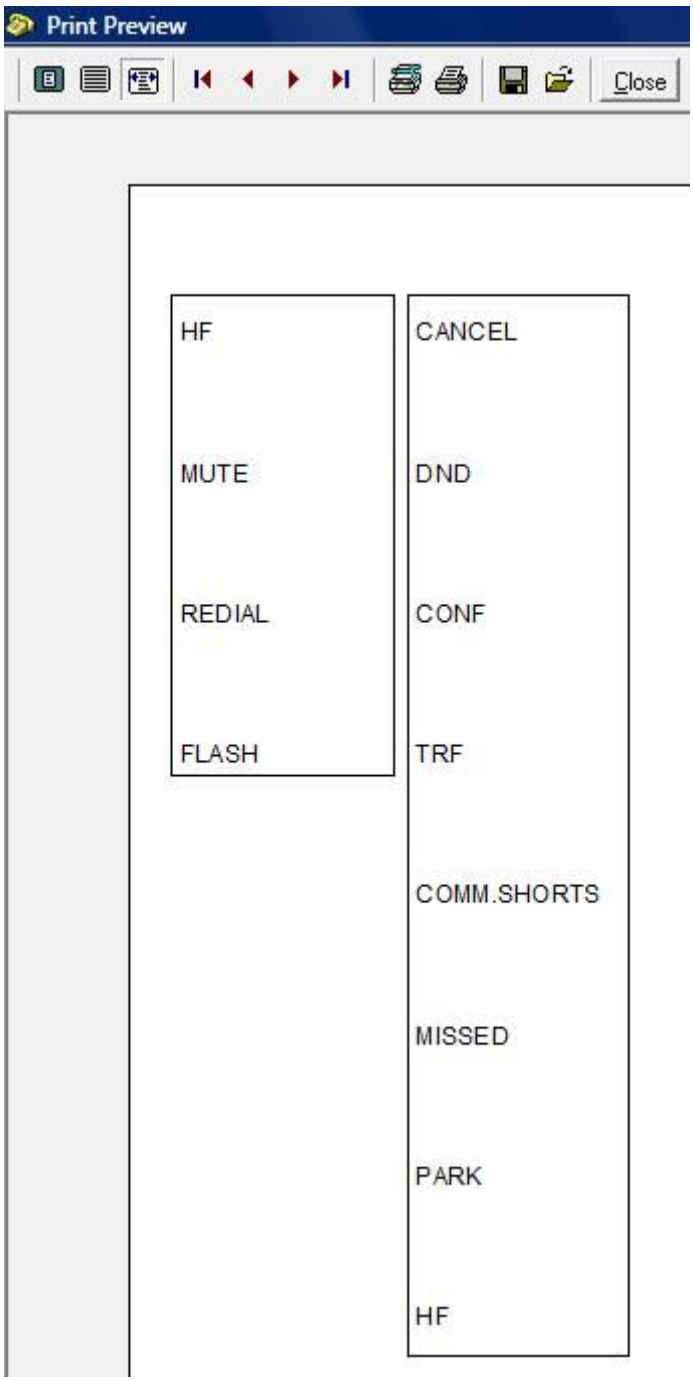
- **Internal call**, the type of ringing from an internal line.
- **External call**, the type of ringing from an external line.
- **Alarm**, the type of alarm call
- **Door communicator**, the type of ringing from the door communicator.
- **Call return**, the type of ringtone if a call is returned.
- **VIP call**, the type of ringtone if an incoming call is from a number in the fast dialling of the given system telephone.

-Back call, the type of ringtone from a line that was blocked by the person being called using the “waiting for free line“ service.
- Special call,

2.3.2. Description of the buttons

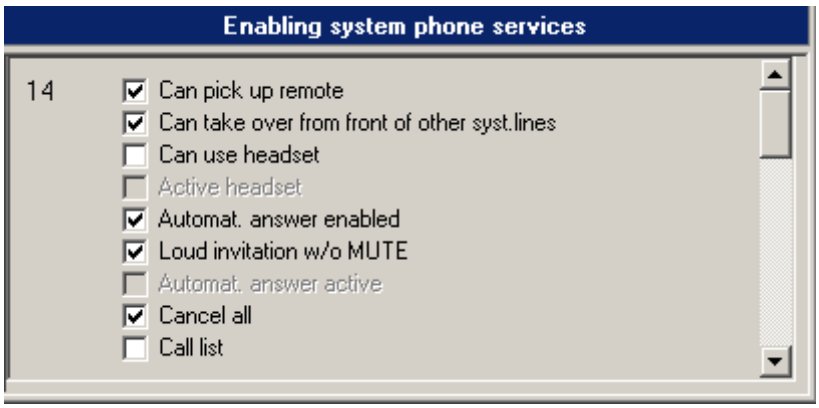
All of the descriptions of the buttons can be formed and then printed out in the environment *PBX configuration tool /system lines /buttons/prints.* After filling in the functions of the individual buttons in MENU *PBX configuration tool /system lines /buttons/buttons* the descriptions are reset, but in *PBX configuration tool /system lines /buttons/prints* it is possible to edit them. The number of letters in a text for individual buttons is given by the width of the individual letters used.

For a “print preview” of the descriptions allow the printing of the individual groups of buttons and press “preview”.



To “Print” the descriptions allow the printing of the individual groups of buttons and press “Print”.

2.3.3. Allowing services



This is where the permission for certain services on the pertinent system phones is done.

- *Can pick up remote* – remote answering is allowed with the voice call service. The speaker of a loud telephone is activated and the microphone is muted.

- *Loud invitation without MUTE* – a similar function as *remote* answering but activated by the microphone.

- *Can take over from front of other system lines* – by pressing a light button it is possible to take over a queued call from another system telephone.

- *Cancel call* through the permission it is possible to refuse a call to an independently ringing system telephone by pressing the Cancel button and then no longer continuing in the ringing table.

- *Call list* by allowing this it enables a printout of the calls to be made. We recommend prohibiting this for system phones that do not support this function (necessary to have at least the 05,12 version of the system phone).

- Can use the headset this permission enables the "Speaker" (loud telephone/headset) function to be controlled through the MENU bar when the headset is connected.

- Automatic answer enabled when activated by the button on the system telephone it enables automatic answering using the "loud telephone" during an incoming call.

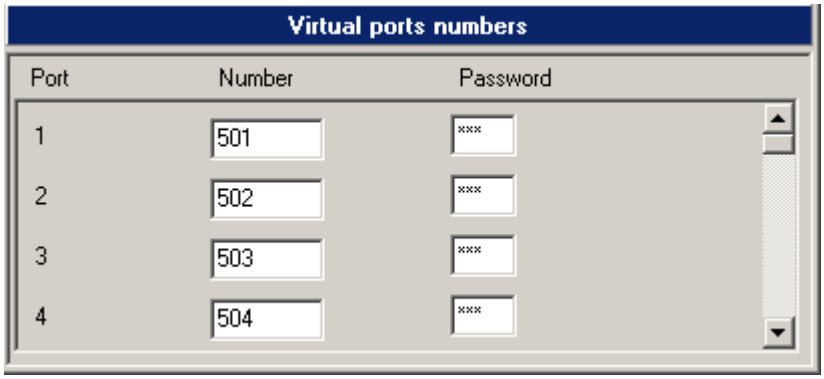
2.4. Virtual lines

2.4.1. Numbering the virtual lines

This item contains a list of the PBX's virtual lines. The virtual line is an extension that does not exist physically. On its account the authorisation, ARS and other parameters can be set for an outgoing call.

A **subscriber number** is allocated to individual ports and can be up to four digits with a four digit **PIN**, used to identify a virtual line. Attention - a collision with the numbering of the other lines and services must not arise.

By right clicking the mouse the ports can be automatically numbered.



2.4.2. Direct selection on the bundles allowed

The same set up and intent as with physical lines.

2.4.3. Tariff credit

The same set up and intent as with physical lines.

2.4.4. Authorisation

The same set up and intent as with physical lines except that here only private authorisations can be set up. It does not distinguish the DAY/NIGHT regime.

2.4.5. Allocating a bundle

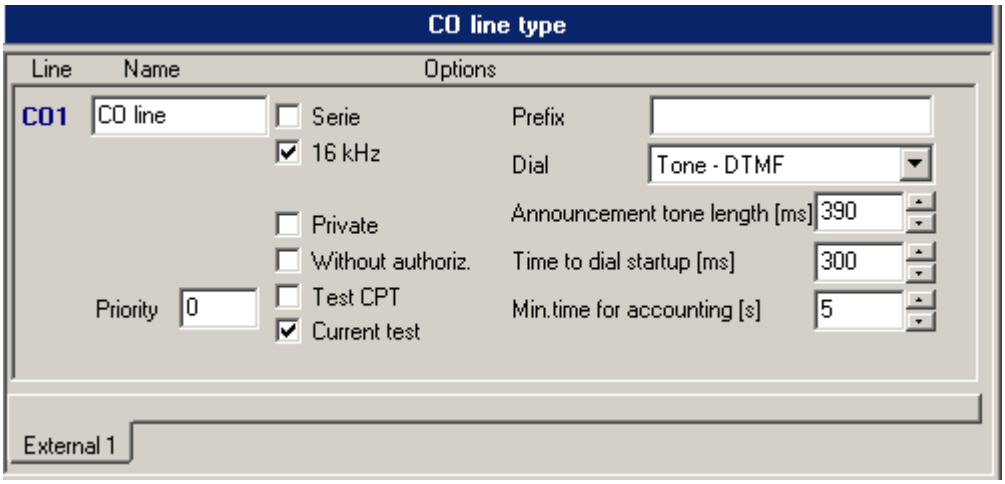
The same set up and intent as with physical lines except that here only allocating the bundle is set up.

2.4.6. Mask for dialling analysis

The same set up and intent as with physical lines.

2.5. External lines

2.5.1. Types of lines



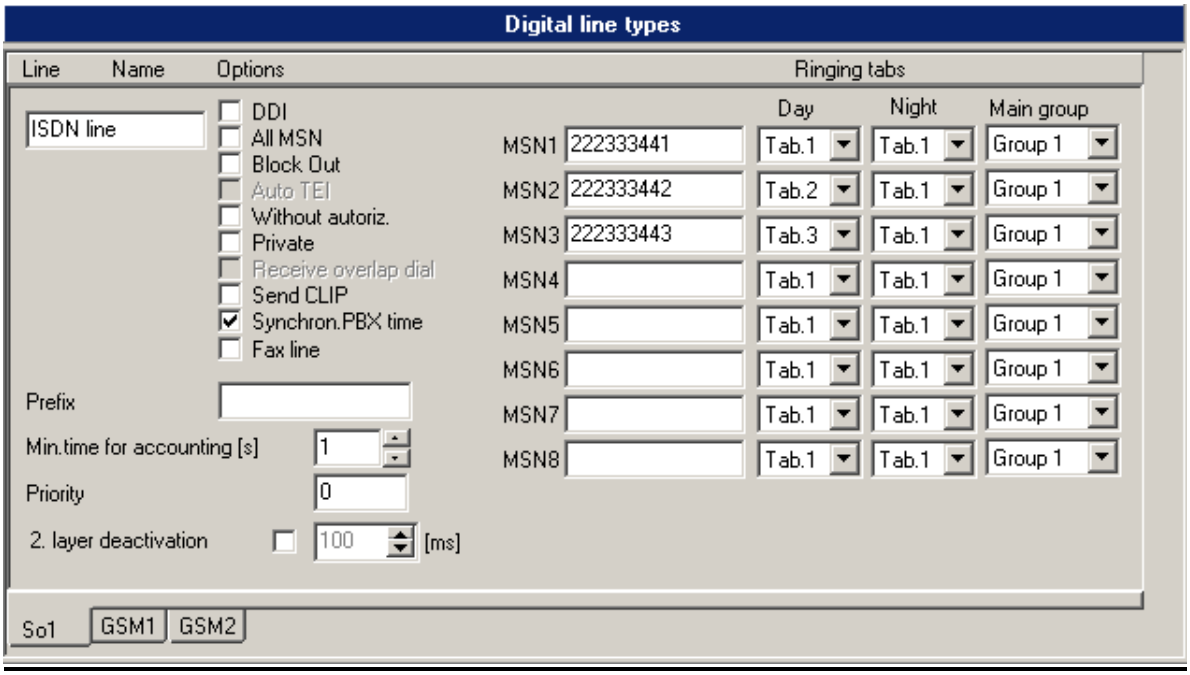
In this table the properties of the individual lines for the analogue external lines is set up

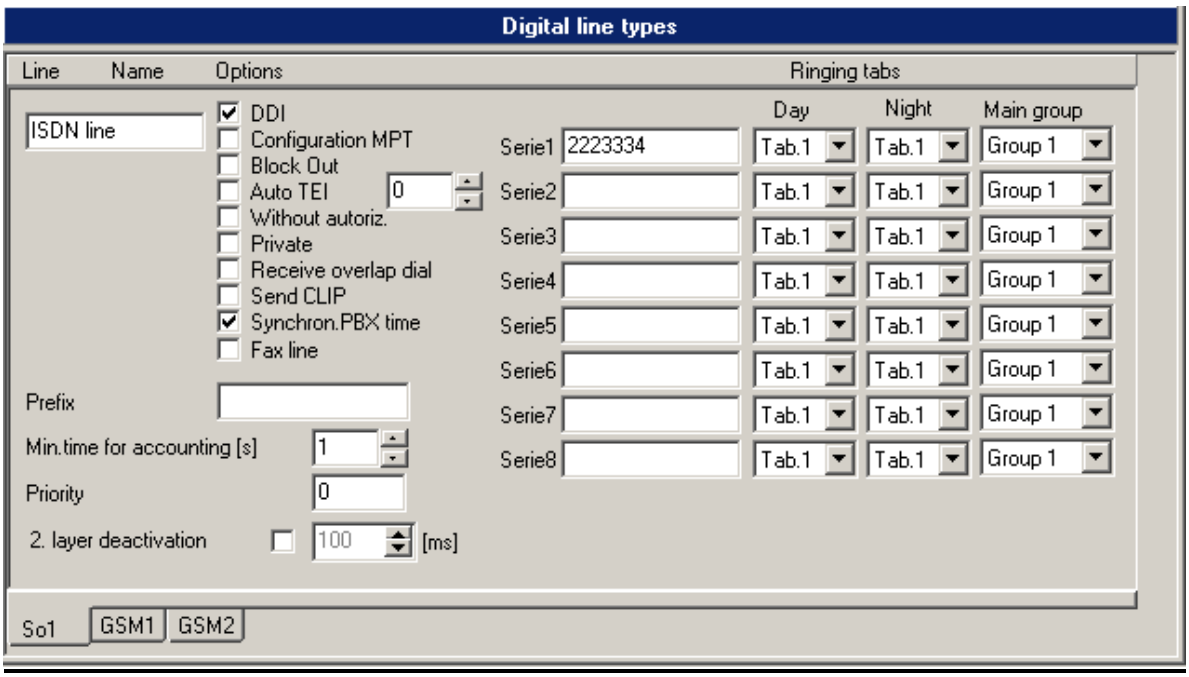
Analogue external line

- **Name –** a maximum 14-character name for the external line
- **Series –** only an incoming external line which has not been tested for correct working and cannot be entered upon in the outgoing direction
- **16 kHz –** an external line with a 16 kHz impulse receiver
- **Private –** a line connected to a private network, which is not charged.
- **Without authorisation –** on this line a check of authorisation is not made
- **Test CPT.** A test for the dialling tone on the external line
- **Current test.** The other option is to test the line current when answering
- **Prefix-** a 4-digit prefix, which is automatically chosen as the first (DID through a superior PBX) after engaging a line. The prefix is not recorded in the charging row.
- **Dialling –** setting the type of dialling for an external line
- **Announcement tone length–** determines how long it must detect an external dialling tone so that it can designate the line as working. If the value “0” is set then the tone is never checked and this line will thus never report a fault.
- **Time to dial startup** determines the minimum time for dialling to be started after the external dialling tone has been detected
- **Min time for accounting –** the period deducted from the call time for temporary charging
- **Priority –** an internal line must have higher or the same priority so that it can use this external line.

2.5.2. Types of digital lines

S_0 line

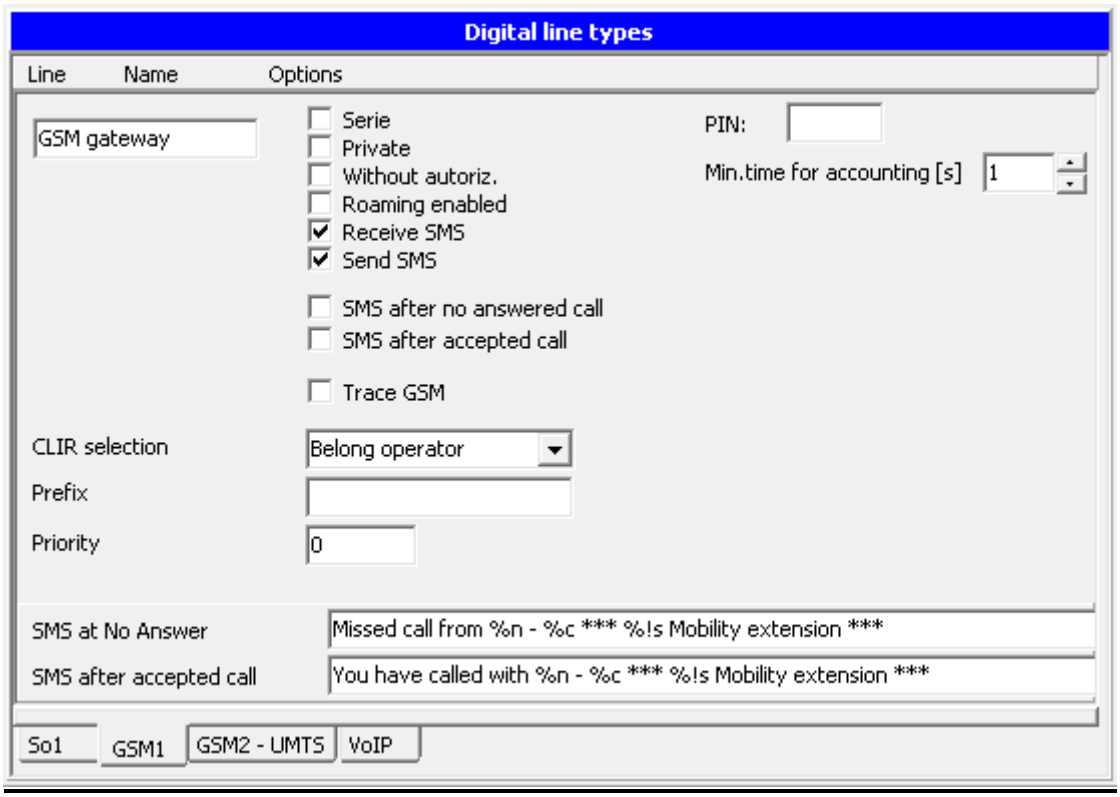




- **Name -** a maximum 14-character name for the external line
- **DDI –** ISDN line in the **PTP** connection
- **MPT configuration** – enables NT1+ analogue telephone equipment to be connected to an ISDN line with a DID.
- **All MSN –** reacts to all incoming numbers by a ringing set on the **MSN1**

- **Block Out –** only an incoming external line and it cannot be entered on in the outgoing direction
- **Auto TEI**
- **Without authorisation –** checks on authorisation not made on this line
- **Private –** a line connected to a private network to which charging is not done
- **Receive overlap dialling –** only for special PTMP applications
- **Send CLIP** – it is possible to forward the caller's CLIP to a private ISDN line.
- **Synchronising PBX time** – during an outgoing call the PBX time is synchronised with the ISDN operator's time (if the operator provides this)
- **Fax line** – labelling an ISDN fax line. Calling is with 3.1 audio signalling.
- **Prefix-** a 4-digit prefix, which is automatically chosen as the first (DID through a superior PBX) after engaging a line. The prefix is not recorded in the charging row.
- **Min. time for accounting –** the period deducted from the call time for temporary charging
- **Priority –** an internal line must have higher or the same priority so that it can use this external line.
- **Series$_{1-8}$ –** fill in the series number or numbers (incoming series number without DID numbers or leave the series empty – if the ISDN network operator only sends a DID number, for precise information ask the operator of the ISDN network)
- **MSN –** the line in the **PTMP** connection
- **MSN$_{1-8}$ –** fill in the **MSN** number 1 to 8
- **Main group** – dividing up the MSN numbers into two groups for the option of switching to the DAY/NIGHT regime and other functions related to group 1 / 2.
- **Deactivating 2nd layer** – by authorising it deactivates the second ISDN layer after setting the time.

GSM line



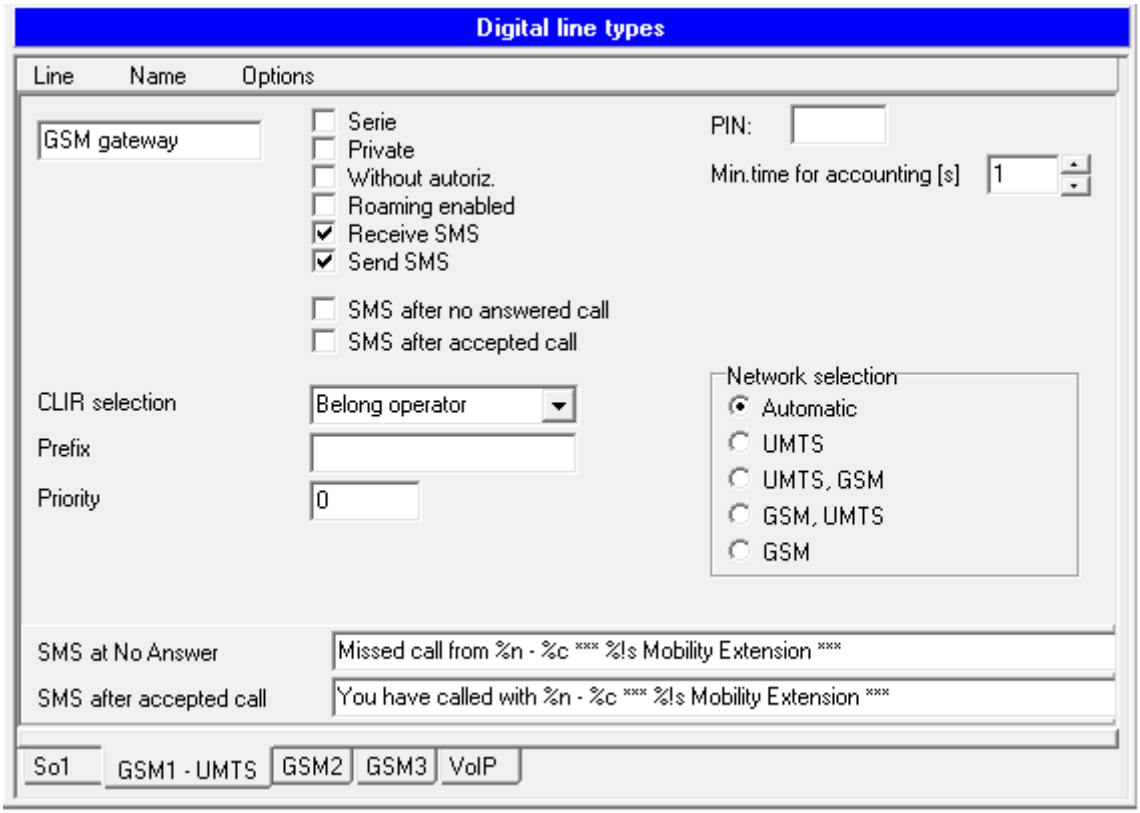
- **Name -** a maximum 14-character name for the external line
- **Series –** only an incoming external line which is not tested for correct working and cannot be entered upon in the outgoing direction
- **Private –** a line connected to a private network, which is not charged
- **Without authorisation –** checks on authorisation not made on this line
- **Roaming enabled** – allowing roaming on the GSM port
- **Receive SMS** – allowing SMS to be accepted
- **Send SMS** – allowing SMS to be sent
- **SMS after no answered call** – the SMS is sent if the called party does not receive the call (the missed call)
- **SMS after accepted call** – the SMS is sent if the called party answers the call (the established call)
- **Trace GSM** – enable logging of AT communication via a GSM module into LogSystem listings. By doing so you make the log heavily loaded, so think twice before enabling.
- **SMS at No Answer** – The text is used in the SMS if the called party does not answer the call (the missed call) (text is common with text for MobilityExtension feature
- **SMS after accepted call** – The text is used in the SMS if the called party answers the call (the established call) (text is common with text for MobilityExtension feature)

%n = Calling party number

%c = Calling party name

- **CLIR selection** – a choice of sending identification – by operator
  - hide number
  - show number
- **Prefix-** a 4-digit prefix, which is automatically chosen as the first (DID through a superior PBX) after engaging a line. The prefix is not recorded in the charging row.
- **Min. time for accounting –** the period deducted from the call time for temporary charging
- **Priority –** an internal line must have higher or the same priority so that it can use this external line.
- **PIN** – a four digit code for accessing SIM cards

GSM linka-UMTS



This setting is for voice parametres GSM-UMTS module to use it for common calling. If you want to use also UMTS DATA (internet connection), you have to set it in the web interface of the VoIP module.

Use your web browser to connect into the the VoIP module. In the Network/Data connection item enter parametres which you get from your UMTS provider.

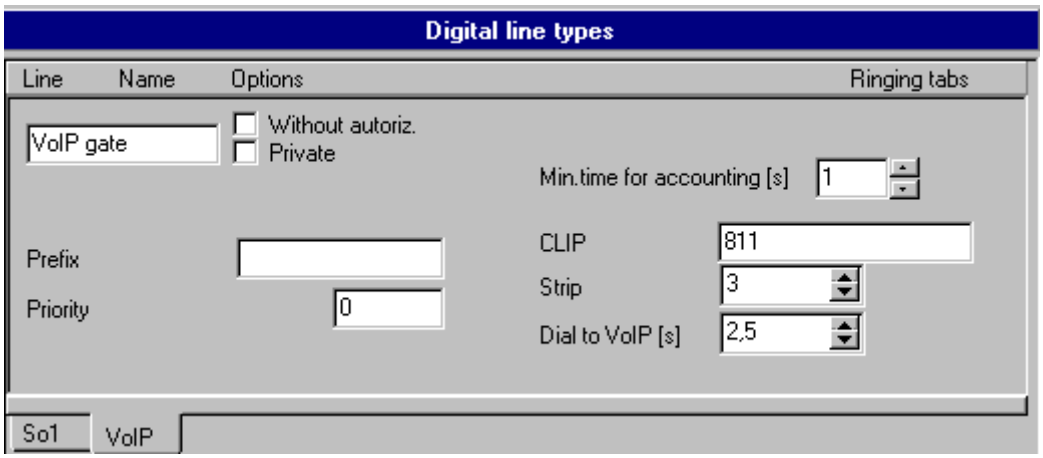
- **Data connection –** enabling of the function
- **Connection provider name –** Informative name, e.g. provider name
- **Dial number (optional) –** data access (optional, made by system).
- **APN (access point) –** login to the provider – parameter requested by provider
- **Initial AT commands (optional) –** not necessary
- **PPP username (optional) –** username for connection to the provider – parameter requested by provider (not always)
- **PPP password (optional) –** password for connection to the provider – parameter requested by provider (not always)
- **Description –** Informative item

Accept settings by clicking at „Modify“ item (Icon – paper with pen)

Example setting of the VoIP module:

You can check successfull connection in menu „Network / Main configuration“. In case of successfull connection you can see there IP of the provider.

VoIP



- **Name –** line name of max. 14 characters
- **Without autoris. –** the authorisation check will not be performed on this line
- **Private –** line connected to the private network, no accounting is performed
- **Prefix –** prefix of up to four digits, which will be automatically called after outgoing occupation of the line as the first (DIN via the superior central). Prefix is not recorded into the accounting line.
- **Minimum time for accounting –** time subtracted from the call time for pseudo-accounting.
- **Priority –** internal line must be of higher priority to use the external line.
- **CLIP** – enter, how should the line be identified in the outgoing direction
- **Strip** – enter number of digits from the incoming identification, which should be subtracted in the input direction in order that the remaining digits are the DIN to the participant of DIN to the ringing table.
- **Dial to VoIP** – maximal period for delay of the next selection into the VoIP line. The time survey is reset after each obtained digit from the internal participant and after its expiring, the dialling mode is changed to the call mode. If it is activated, the expiring of the period and end of the dialling is announced by a short beep. Any other dialling is considered a service into the call.

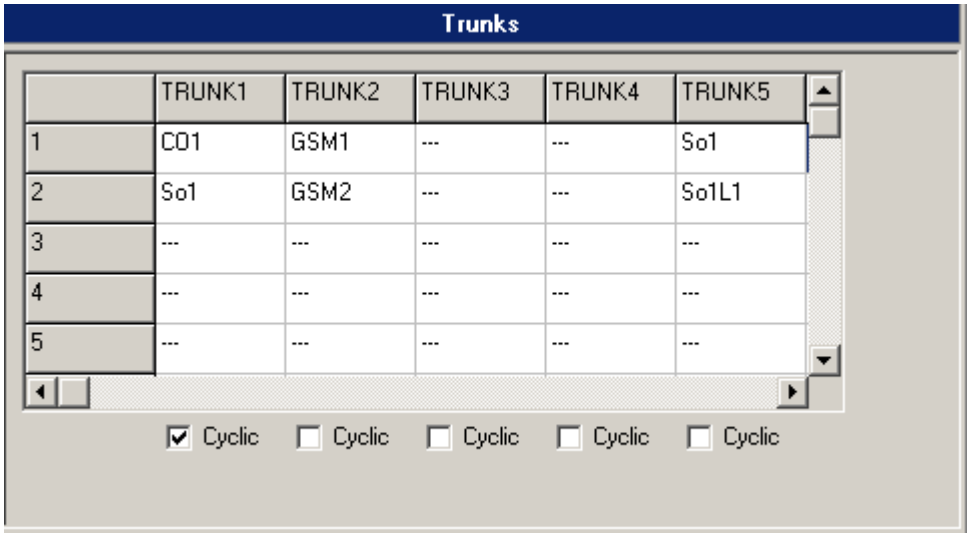
2.5.3. GSM Services

Set the GSM ports for which the GSM CLIP function is to be enabled.

- **CLIP In** – enter the expected incoming GSM identification format;
- **CLIP Out** – enter the expected outgoing GSM identification format.

These services are rendered by some GSM providers only.

2.5.4. Trunks



The table serves for allocating individual external lines to a bundle. The maximum number of external lines in the bundle is **24**. There are 16 bundles. Every external line may be in one or more bundles at once. The first eight bundles can use the service **Internal call through a bundle**. A cyclic start up for an external line can be used in the bundle.

2.5.5. Trunks for accessing ringing

The table serves to allocate individual external lines to a bundle for using the service **accessing ringing from a bundle**. Every external line can be in one or more bundles at once.

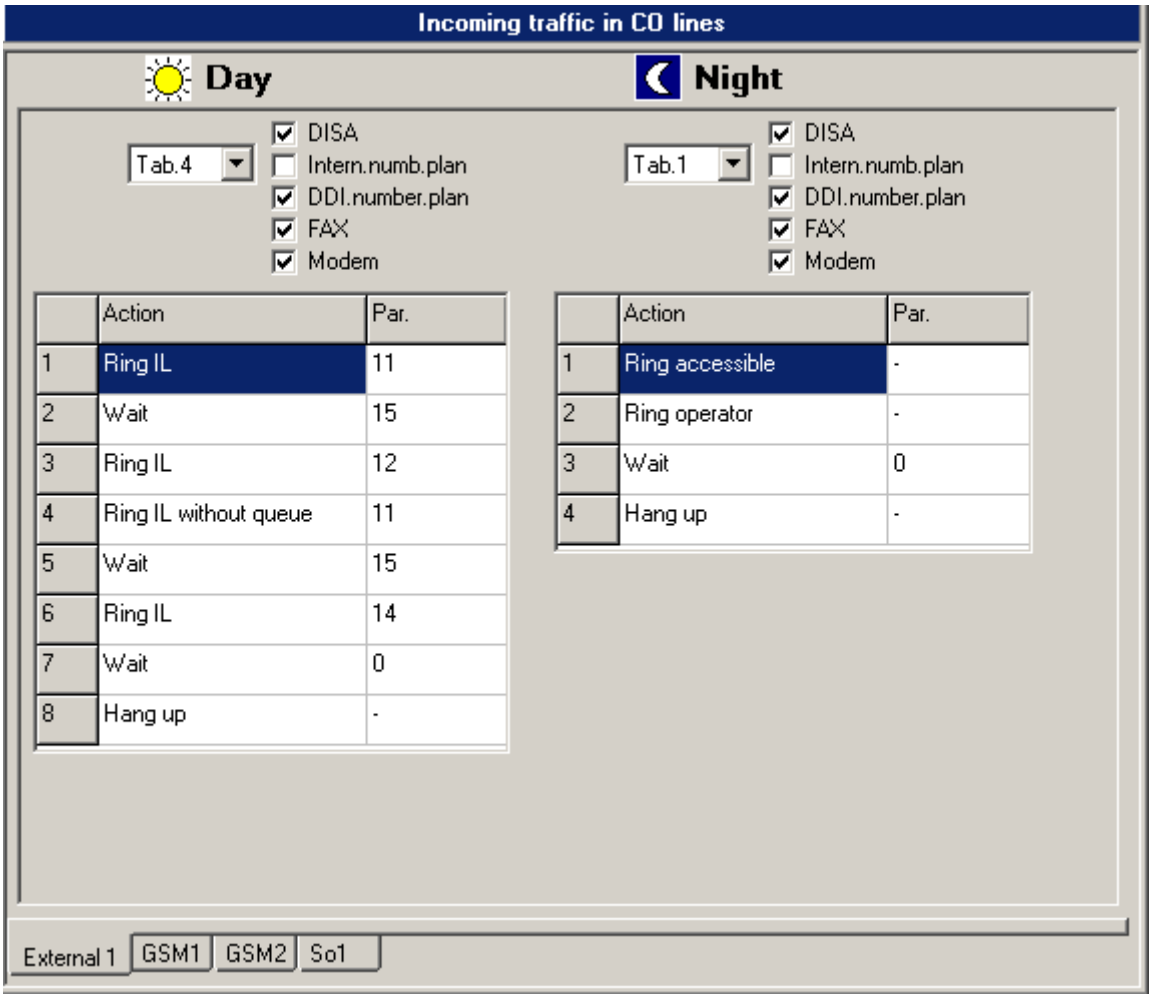
2.5.6. Ringing

For every external line you can set up the same or a different ringing table for the DAY and NIGHT regime. The tables are general and the same table can be used for various ports.

- **DISA –** when setting it is possible to call the DTMF by selecting an internal subscriber
- **Internal numbering plan –** a number received during DISA DID will be compared with the subscribers' internal numbering
- **DDI numbering plan –** the number during DISA DID will be compared with the subscribers' DID numbering
- **FAX –** when using the DISA command in the ringing table the fax 1100Hz introductory tone will be detected
- **Modem –** when using the DISA command in the ringing table the modem 1300Hz introductory tone will be detected

Here you can select where the DISA is entered in the ringing table, so that after an incoming ringing on an external line it takes into account FAX or Modem tone in the DTMF selection and connects to the declared internal line in accordance.

Ringing from external lines can elicit a maximum of 500 events.



2.5.7. Ringing table

One of the following actions can be programmed in each row:

Action	Parameter	Description
Ring VL	VL number or SYS	Ringing for a specific number
Ring VL without a queue	VL number or SYS	Ringing for a specific number. If engaged then it does not form a queue. If only one ringing then it rings on.
Ring VL without a queue or jump	VL number or SYS	Ringing for a specific number. If engaged then it does not form a queue and carries out the command two rows below.
Ring VL without a queue and jump	VL number or SYS	Ringing for a specific number and jumps two rows down. If engaged then no queue formed and carries on in the ringing table.
Ring VL and set for voicemail	VL number or SYS	Ringing for a specific number and thus identifying itself even in a voice system (if the ringing is forwarded).
Don't ring VL	VL number or SYS	Ends ringing to a specific number.
Ring accessible	None	Ringing accessible internal and system lines
Ring accessible cyclically	None	Cyclic ringing accessible internal and system lines
Ring accessible linearly	None	Ringing the first of the accessible internal and system lines
Ring accessible cyclically no queuing	None	Cyclic ringing accessible internal and system lines. If it is engaged then it does not form a queue. If only one ringing then it rings on.
Ring accessible linearly no queuing	None	Ringing the first free accessible internal system line. If it is engaged then it does not form a queue. If only one ringing then it rings on.
Ring dept. linear.	Departments 1 to 8	Rings first free number from the department
Ring dept. cyclic.	Departments 1 to 8	Cyclically rings first free number from the department
Ring entire dept.	Departments 1 to 8	Rings entire department
Ring dept. linear. without queue or jump	Departments 1 to 8	Rings first free number from the department. If internal lines engaged then it does not form a queue and carries out the command two rows down.
Ring dept. cyclic. without queue or jump	Departments 1 to 8	Rings another free line from the department. If internal lines engaged then it does not form a queue and carries out the command two rows down.
Ring entire dept. without queue or jump	Departments 1 to 8	Rings entire department. If internal lines engaged then it does not form a queue and carries out the command two rows down
Ring operator	None	Rings the operator
Ring voicemail	VL number or SYS	Rings the voice mailbox given by the parameter
Action	**Parameter**	**Description**
Don't ring	None	End all ringing
Wait	Time - from 1 to 999 [s]	Wait for the set length of time, 0 = unlimited wait

		or until the incoming ringing from an external line
DISA	Number of voice from 1 to 10 = 0 Or nothing	Stops previous ringing to internal lines and takes an external line to wait for a fax or modem call or DTMF number
DISA End	None	Ends dialling detection
Voice	Number of voice from 1 to 9	Plays the voice
Select tone	A selection of tones	After lifting an external line with the DISA service activated a different tone can be imposed (for special applications)
Table	Number of ringing table	Ringing tables can be catenated
Automatic operator	Automatic operator's number 0-8	Rings a given automatic operator voice system. If module not working then it carries on with next row in table.
Connect remote access	None	Rings the remote access modem in the PBX. If the module is not installed then it carries on with next row in table.
Transit DID	VL, SYS number or virtual	Carries out a start up on the bundle for outgoing lines to the VTS by parameter. This allows an outgoing call to be made from within the PBX on the line account given by the parameter.
Call fast dialling	The number or name of the joint fast dialling	After an incoming external call it dials the given fast dialling. Which line the call will be charged to and which lines it will go to the VTS is determined in the settings External lines/Identification for transit calls.
Hang up	None	Hangs up on an external line and ends all actions. It is in the last row and cannot be removed.
Go to line	Row number	Jumps to the selected ringing table row.
Announce through broadcast	Voice message	Connects the AUX assigned to the **Broadcast** service, plays the voice message and the AUX Broadcast remains connected until hang-up or the **Terminate broadcast** service activation.
GSM DTMF test	None	While calling to a module equipped with DTMF detectors (designated as OL2GSMD, OLGFAX at 2N), you can dial the DTMF characters during ringing that are displayed on the PBX.

Attention: *The total number of rows in all of the ringing tables cannot exceed 500.*

Here are some working examples for illustration.

Example 1: During incoming ringing it has to concurrently ring lines 13, 14, 15 and 16 and all lines that have ringing from external lines accessible. All lines ring at the same time until one of them takes the call or the call ends externally.

Action	Parameter
Call accessible lines	
Call VL	13
Call VL	14
Call VL	15
Call VL	16
Wait	0

Tab. 1

Example 2: During incoming ringing it starts ringing line 12, after 10 secs it stops on line 12 and starts on line 14. After another 5 secs it transfers ringing to lines 13 and 15. Then all 3 lines ring until one of them takes the call or the call ends externally. Of course it must ring the accessible line first.

Action	Parameter
Call accessible lines	
Call VL	12
Wait	10
Don't call	12
Call VL	14
Wait	5
Call VL	13
Call VL	15
Wait	0

Tab. 2

Example 3: During incoming ringing it rings the group linearly. Again we bear in mind the lines that have ringing from external lines accessible.

Action	Parameter
Call accessible lines	
Call dept. linearly	5
Wait	0

Tab. 3

Example 4: During an incoming call it rings group 3 cyclically. Accessible lines don't ring. If, after 60 secs, no one answers it ends on the voice mail of subscriber 11.

Action	Parameter
Call dept. cyclic	3
Wait	60
Don't call	
Call voicemail	11
Wait	0

Tab. 4

***Example 5**:* During an incoming call after 4 secs the external line is picked up, it waits for the DISA or fax dial tone with the voice signal number 1. After it subsides it waits another 6 secs and if the DID does not come then it rings for 20 secs on lines 12 and 13. Afterwards it forwards the call to the operator and if no one takes the call after 30 secs it records voice 3 (e.g. you will be connected to the answer phone), it connects to the answer phone on line 19 and if the answer phone does not accept the call then it hangs up after 90 secs.

Action	Parameter
Wait	4
DISA	1
Wait	6
DISA End	
Call VL	12
Call VL	13
Wait	20
Call operator	
Wait	30
Don't call	
Voice	3
Call voicemail	11
Wait	90
Hang up	

Tab. 5

Example 6: Delayed DISA. During an incoming ringing it rings for 20 secs on the accessible lines and line 19. If no one picks it up, then it picks up the external line then it awaits the DISA or fax DID with the voice signal number 1. After the signal subsides it waits 6 secs and if a DID does not come in that time it rings line 12 and the operator for 30 secs. If no one takes the call in this time it plays voice 2 to the subscriber and hangs up.

Action	Parameter
Call accessible lines	
Call VL	19
Wait	20
DISA	1
Wait	6
DISA End	
Call VL	12
Call operator	
Wait	30
Don't call	
Voice	2
Hang up	

Tab. 6

Example 7: Using CLIP support, ringing from an external line is directed to internal line 15. The time it rings on line 15 is not influenced by the ringing table and is controlled by the settings in line 15. For this case it is most important to prearrange the command Wait with the parameter 2 secs before the actual command Call VL with the parameter 15.

Action	Parameter
Wait	2
Call VL	15
Wait	0
Hang up	

Tab. 7

IMPORTANT!!
When using the DISA services in the ringing table then the row Wait 0 is not allowed to occur !

2.5.8. Global data

see *Global data for external lines.*

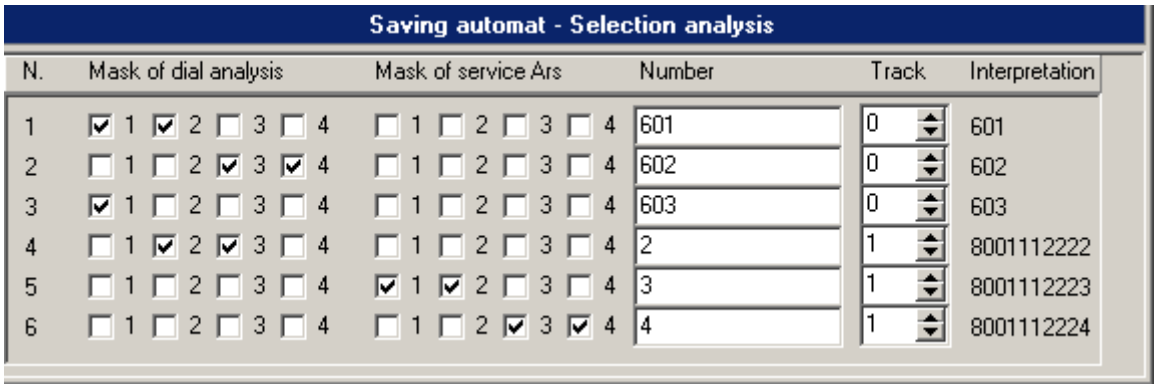
2.5.9. Groups and switching

Here external lines are allocated to one of the two main groups depending on which decides whether it is the DAY or NIGHT regime and according to this a ringing table is selected.

2.6. Saving automat

After dialling the service “access to an external line” none of the external lines are activated. For lines that use the LCR, the PBX simulates an external dialling tone and awaits the next number dialled. According to an analysis of the next number dialled the PBX determines the connection and carries out the connection. For an internal subscriber entry to the LCR is obligatory, possible or prohibited (as determined in the Authorisations Table). At the same time it is possible to use up to 4 analysis masks and these being in combinations that allow or block the analysis of dialling depending on the individual rows of the **Dialling Analysis** table.

2.6.1. Dialling analysis



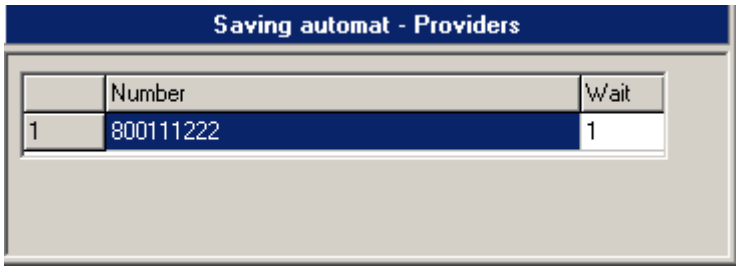
In the rows of this table the number dialled to an external network from a subscriber who has ticked the choice **Use the LCR** is compared with the numbers entered here. This comparison eliminates the rows that are not in agreement in at least one of the four masks for dialling analysis for an internal subscriber and the **Rows of the dialling analysis.** The result is a route number (routes ranging from 0 – 150), though which the outgoing call will be dispatched.

In the table rows it is also possible to set up a mask for choosing the **ARS 1-4** service. These services are primarily used for access to private networks.

The next chapter gives some illustrative examples.

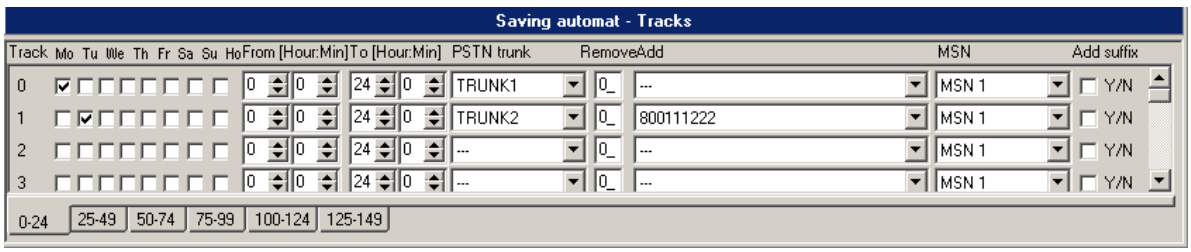
2.6.2. Routes - providers

Here routes are allocated that can be used by the external line bundles to make a connection. They are always selected in the order of the first external line from the bundle. At the same time this is where it is defined how many numbers from the number called are subtracted and what numbers are prearranged – or which operator will be selected for providing a least cost route.



It is also necessary to enter on what day and hour the given route is to be used. The selected bundles use it independently of the programmed bundle selection for the internal subscribers in the authorisation table. It is possible to take away 14 numbers at most and the number of automatically prearranged numbers is not allowed to exceed 30. The symbol “T” means crossing over to tone dialling, a comma (,) means a delay in dialling and “W” means switching to tone dialling and waiting for log in (possible for ISDN, int. GSM or analogue external lines with a tariff impulse, E&M), if a log on does not come then it carries on with the DTMF dialling after the set time.

For each route, if ISDN lines are used in the outgoing bundle, then it is possible to assign identification using the MSN numbers identically as when setting the services and private MSN numbers for the internal lines. It is particularly beneficial to use this for outgoing calls to private circuits.


Saving automat - Tracks
Track Mo Tu We Th Fr Sa Su Ho From [Hour:Min] To [Hour:Min] PSTN trunk RemoveAdd MSN Add suffix
0 TRUNK1 MSN 1 Y/N
1 TRUNK2 800111222 MSN 1 Y/N
2 MSN 1 Y/N
3 MSN 1 Y/N
0-24
25-49
50-74
75-99
100-124
125-149

Example 1: How to learn that during outgoing calls to a number **starting with 0603 route 0** is used, which goes through the PBX’s **GSM module** of the operator Paegas.

The dialling to the Paegas mobile network given above has a predetermined **route 0**. This, for instance, is allocated to **bundle 5** in which there is the external line with the **GSM module.** None of the numbers have to be added or subtracted as the code **0603** is still in the memory (nowhere was called, it was only used for decoding to allocate the route) and after being **picked up** the port in **bundle 5** calls to it. **The next number** dialled by the internal subscriber is **automatically filled in** thus achieving **cheap** calls to a mobile phone from an internal subscriber though the GSM module.

Example 2: Our PBX is in Prague. If, for instance, the outgoing call starts ***05 333666***, which is the number of the firm’s branch in Brno, which also has the module for GSM

telephony, we can determine that this Brno number uses, of instance, ***route 1***, which also goes through a ***GSM*** module and totally replaces the number dialled with the GSM number of your branch.

Here it is predetermined that the number uses ***route 1***. ***Bundle 5*** (an external line with the GSM telephony module) also uses this, but here it is necessary to ***take away*** all ***8*** numbers (in the box **Subtract** the number 8 is entered) and ***replace*** them with, for example, the number ***0603234567*** (this number will be entered in the column **Add**), because this number is the number of the module for the GSM telephony of your Brno branch. This again gives a cheaper connection from Prague to Brno than through the land line network.

Example 3: Imagine that the fax line also uses the LCR, but because faxes can't be sent through GSM, it is necessary to prevent this. This can be done, for instance, such that the fax line will have a different dialling analysis mask ticked than the row with the number of the Brno branch, which alternates the connection through GSM. Thus we achieve a state in which the fax line cannot "find" a cheaper exit for the Brno branch and returns from the LCR to make the connection through the standard public network.

2.6.3. Dialling analysis mask

This serves to limit combing through all the rows when calling with the LCR. The **Route Analysis** row will only be searched through if the dialling analysis mask is also ticked for an internal line that is seeking a cheaper route.

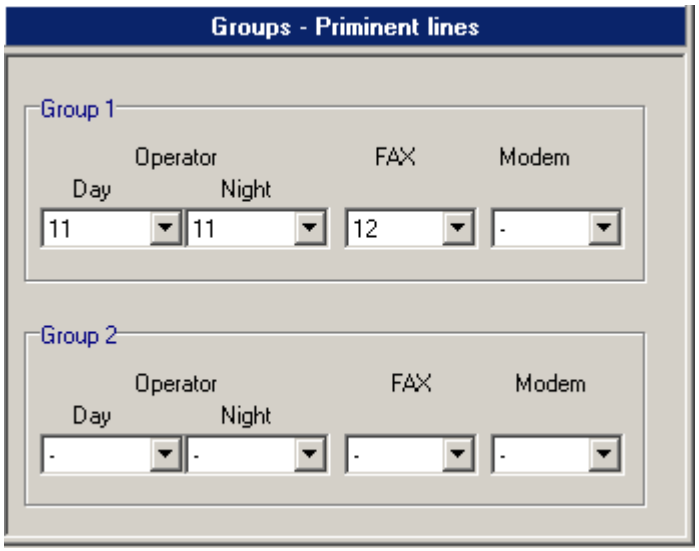
Example 1: Two modules for GSM telephony with the operator EuroTel are installed in the PBX. One belongs to one firm the other to another. In the **Route** Analysis table one row will be entered with the number 602 aiming at bundle 5 (the first module for GSM telephony) and, for example, the dialling analysis mask has 3 ticked. Another row will also have the number 0602 entered but be directed to bundle 6 (the other module for GSM telephony) and the mask will have 4. Afterwards the internal subscribers of the first firm must have mask number three ticked and they are not allowed to tick mask 4. In contrast subscribers in the other firm must have the number 4 ticked in the dialling analysis mask and are not allowed to tick 3. This ensures that each firm will only go through its module for GSM telephony and it won't contain the neighbour's. It is also possible for the director to have both 3 and 4 ticked for the dialling analysis mask on his lines, which for an outgoing call to GSM EuroTel broadens the opportunity of entering both modules for GSM telephony. And why not when the PBX can charge these calls separately for internal subscribers.

Example 2: With the help of the LCR the PBX can save not only money but also time. Imagine a certain department of people that are allowed to call an external network for work on numbers (DID) that are often repeated. The DID of the affiliated branches will be entered in the LCR's **Route Analysis** rows, and these calls will have the route with the largest available amount of external lines in the bundle or bundles. If we set the internal lines of this department so that for a normal outgoing call (not through the LCR) then they will only have one bundle that doesn't have many external lines and they have the **Use LCR** ticked, then the frequently dialled numbers of the affiliated firms are easily called through a well sized bundle in the routes of the LCR, but for unknown DIDs it will be more difficult as only a very limited number of external lines will be available for this call.

Example 3: Other authorisation. If an internal line was set to **Obligatory use of LCR**, then the numbers of frequently phoned business partners could be entered into it. Then an internal subscriber calls these DIDs through the LCR route but they cannot call to other numbers in the public network because if a chosen number is not found in the LCR table that is obligatory for the subscriber, then the subscriber gets the engaged tone.

2.7. Groups

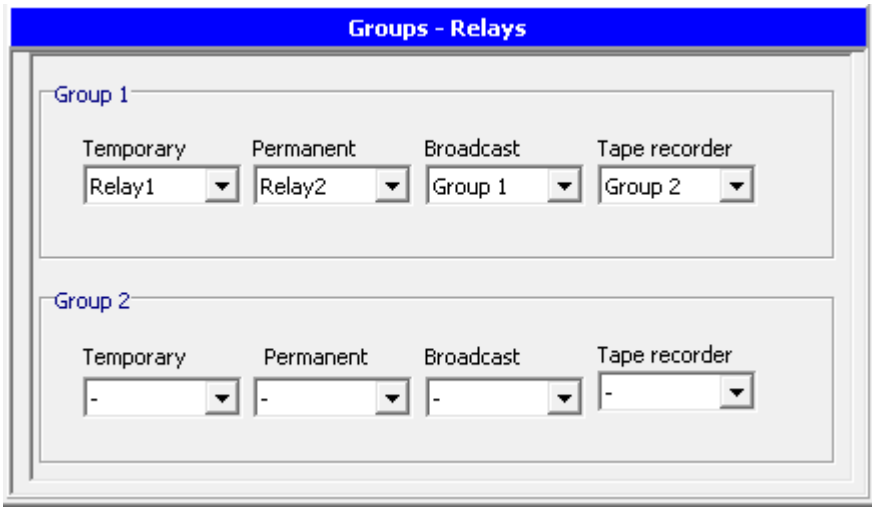
2.7.1. Designating lines



It is possible to split subscribers into two groups on the PBX. For each main group it is possible to define the following privileged subscribers:

- Day - the number of the subscriber working as the operator for the DAY regime
- Night - the number of the subscriber working as the operator for the NIGHT regime
- Fax - the number of the subscriber where there is a fax for a group
- Modem - the number of the subscriber where there is a modem for a group

2.7.2. Switches, Broadcast, Tape Recorder



Enter the port relay for *Activate temporary/permanent switch* for groups PBX 1 and 2.

- Temporary – activate the switch temporarily = default #85 (the relay is closed until the dialling timeout expires = 15s, the activation is terminated immediately by hang-up). If this item is empty, you can enter the required port relay after dialling the service.
- Permanent – activate the relay permanently = default #83 (the relay is closed permanently and can be opened using the *Deactivate permanent switch* service = default #84). If this item is empty, you can enter the required port relay after dialling the service.

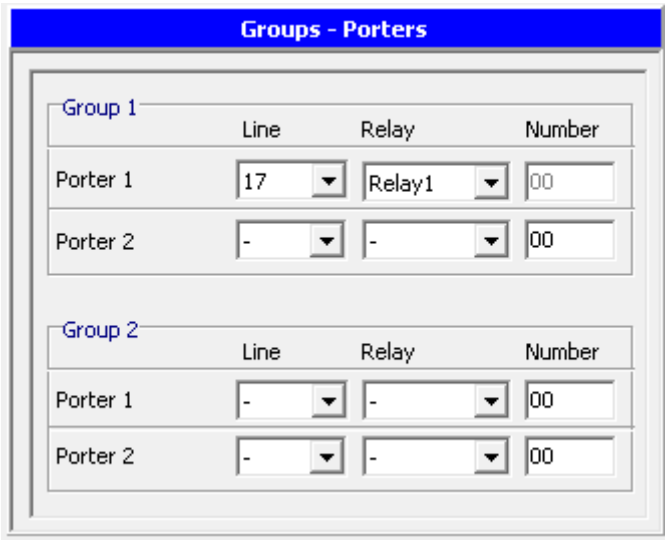
Assign AUX group to Broadcast and Tape recorder

- Broadcast - the selected AUX ports of the selected AUX group will be used for the *Broadcast*=default #20 service at relax and for the *Announce through Broadcast* service in the ringing table.
- Tape recorder - the selected AUX ports of the selected AUX group will be used for the *Switch on tape recorder*=default #20 service during a call and also automatically with every call of the extension for which the *Switch on tape recorder* service is selected in the *Subscriber lines/Service enable /Activation* menu.

Assign the AUX ports into groups in the *Global data/AUDIO groups* menu.

2.7.3. Intercoms

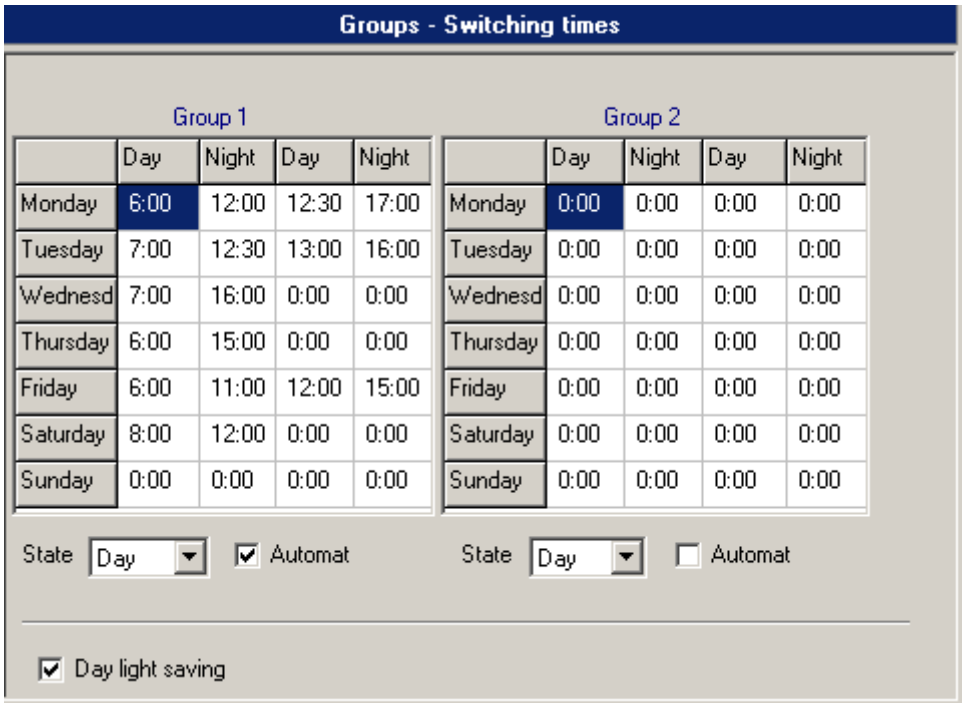
This table determines the intercoms' location on the internal lines. It is possible to set two intercoms for each group. Ringing from this line is characterised by being different from the usual ringing.



The selection **Number** enables a DTMF number to be sent to the intercom (e.g. 00 for *2N*® – INTERCOM.) to open the door.

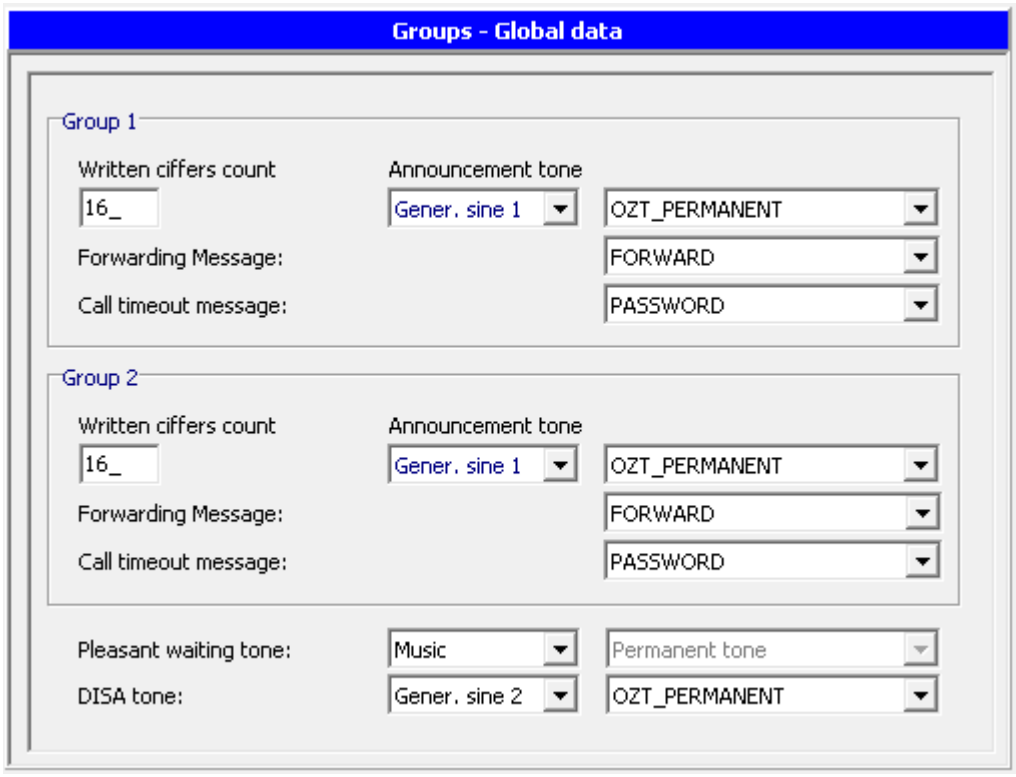
If the door communicator is not equipped with a door-opening relay, use the module relay and select the *Activate relay* service for activation.

2.7.4. Switching time



In this menu it is possible to set two times for automatically switching to the DAY and NIGHT regimes for each group and for every day. Ticking the box **Automatic** gives automatic switching at the requisite times. Not entering anything for the time means that there is no switching. So, for instance, leaving out one box for DAY and one for NIGHT only ensures one cycle of automatic switching during one day. Ticking the box **Automatic** gives automatic switching at the requisite times. The selection **State**, in which the choice of either DAY or NIGHT is entered, ensures the PBX is started in the given regime if automatic switching is not used. The choice **Change to summer time** ensures that the PBX automatically switches over to "summer or daylight saving time".

2.7.5. Global data



The choice **Number of registered figures** determines how many numbers are entered into the charging rows of work calls. Only the first five numbers of private calls are automatically displayed thus this choice has no effect on them, unless it is not set otherwise in the Global data – charging.

The choice **Dialling tone** enables the type of dialling tome to be selected from the constant type and the "Morse" type and the type for calling to a state line. The other types of tones are only for special applications. As a rule it is not necessary to make changes. If you wish to do so, then first select the tone source and then the cadence.

Music to make waiting more pleasant. Defines the source of music used during both internal and external waiting. Other types of tones are only used for special applications.

The voice at the start of redirecting is used at the start of being redirected to the VTS.

The voice when exceeding the call period during redirection is used as a notification that the call must end during redirecting, however at this moment it is possible to extend the call by 1-10 minutes before the voice gives the notification again by selecting DTMF (1-10) and selecting the call period.

Examples of voice:

The voice at the start of the redirecting

"Your call will be redirected"

The voice when exceeding the call period during redirection

"You have a limited time for this call, through a tone selection you can extend the call by one to ten minutes, otherwise your call will be ended"

DISA tone

The tone that is used when calling DISA, if a voice is not played to the caller.

Attention: *If you want to use modem detection on an external line where the DISA is activated without a voice, then you are not allowed to use a continuous dialling tone.*
These voices they are not recorded as standard.

2.8. Numbering

2.8.1. Internal lines

See *Internal lines /Numbering*

2.8.2. Services

- Table for programming the numbering plan of the PBX's services.

2.8.3. DID for the ringing table

Enables an external subscriber to call through to the ringing table during a DID. If the DID number is not in agreement with the subscribers' DID or internal numbering and agrees with the number in the DID table for the ringing table then the ringing table is used, here set up with regards to the DAY and NIGHT regime. Enables the ringing table to be set up during DID (ISDN – Point to Point type, DISA).

2.9. Operating parameters

2.9.1. Internal lines

2.9.1.1. Redirecting

This table clearly displays all of the redirecting for all of the internal subscribers.

2.9.1.2. Tariff credit

The item **Prepaid** serves for assigning the maximum number of 16 kHz impulses received. When the assigned number is reached the call automatically ends. A new call can be made as soon as a new credit programme is topped up.

2.9.1.3. Blocking

Here it is possible to easily check or set blocking for each internal line including its password and not stop the service.

2.9.1.4. Alarms

Set up an "alarm" on each line. It is possible to choose between "Single" or "Repeating".

2.9.1.5. Account statistics

Clearly displays the costs of the internal lines from outgoing calls through the external lines that have tariffs. An update is available during each data reading from the PBX.

2.9.2. External lines

2.9.2.1. Making accessible

The table has as many columns as there are internal subscribers and as many rows as there are external lines. Thus at intersections it is possible to tick or end an internal subscriber's access to each external line.

2.9.2.2. Tariff credit and Blocking

The item **Prepaid** serves for assigning the maximum number of 16 kHz impulses received. When the assigned number is reached the call automatically ends. A new call cannot be made through this external line. Credit must be topped up through a programme.

The item **Blocking** enables an external line to be blocked for example during a breakdown.

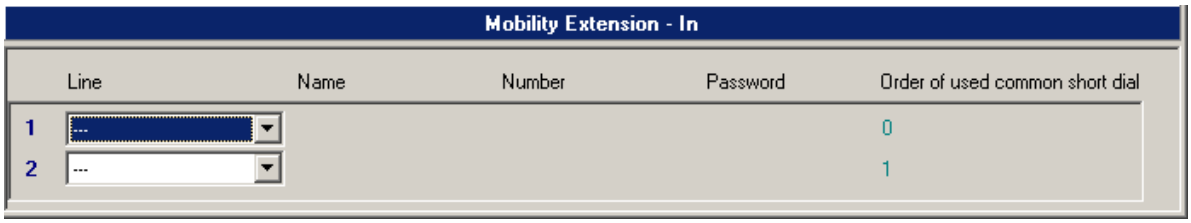
2.10. Mobility Extension

Mobility Extension function enables telephone stations (preferably GSM telephones to share internal line services (switching, dialling VTS,..) outside the PBX.

Primary setup is made by the PBX administrator.

2.10.1. Mobility Extension IN

Here you set up the conditions for Mobility Extension in the incoming direction from the external telephone number (preferably from a GSM telephone).



Enter:

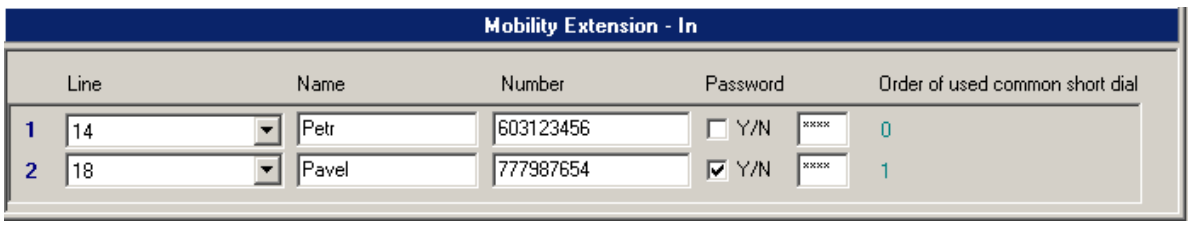
Line - internal that should have the Mobility Extension service (sharing identical services with an external telephone number) activated. One internal line has one external telephone number allocated. After being allocated in one of the menu lines offered (according to the ME license inserted) it will not be displayed in the following ones.

Name - Name of the external telephone number (enter the name without diacritics). The name is shown on the display of the system telephone.

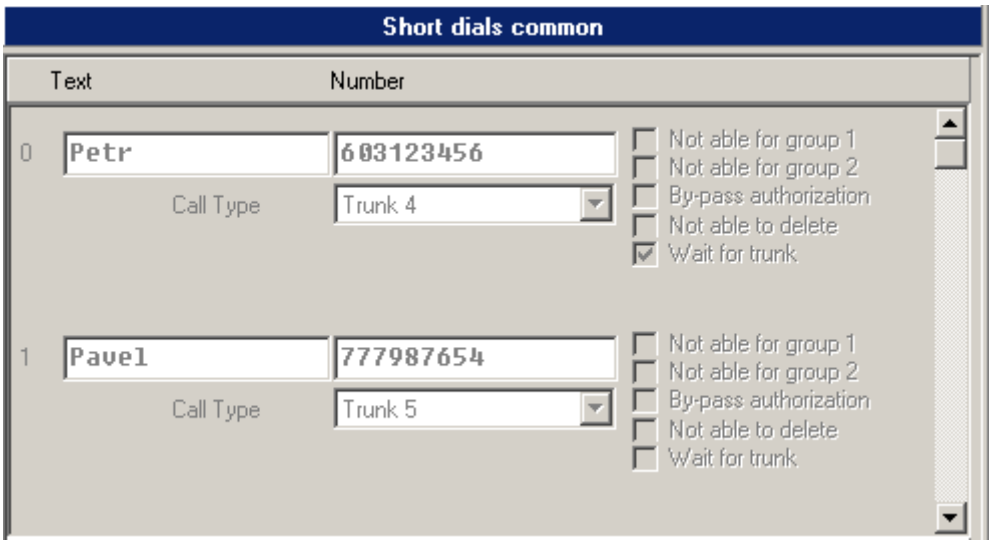
Number - external telephone number.

Password - personal password of the allocated internal line. It is the same as the password set up in the "*Internal lines/Operating parameters/Blocking*" MENU. If it is allowed to enter a password it is always necessary, when calling from this external telephone number, to enter the password first and then further dialling is possible. The user is asked to enter the password with the following message: *"Enter password"*.

Order of used common short dial - after entering the Name and Number, a new common quick dialling is established, whose ordinal number is displayed. Always the first free common quick dialling is offered.



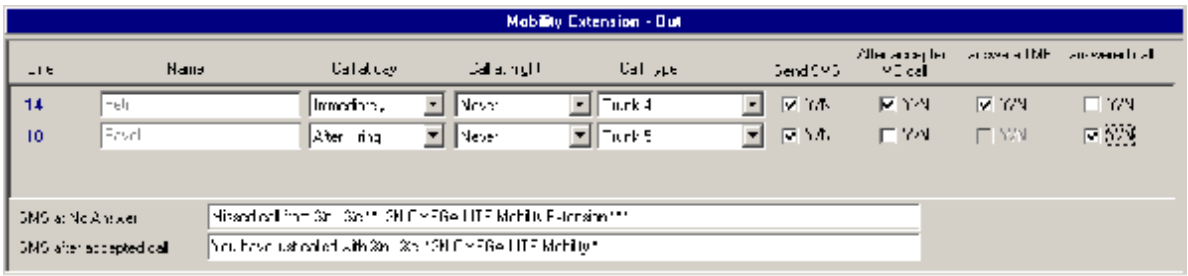
A common quick dialling established in the Mobility Extension IN/OUT environment can only be edited here. Direct editing in the *"Common quick dialling"* MENU is block for such dialling.



2.10.2. Mobility Extension OUT

Here you set up the conditions for Mobility Extension in the outgoing direction, the allocated internal line rings.

Editing the settings for the outgoing direction is only possible if Mobility Extension IN is entered. Otherwise no MENU is offered.



Line and **Name** of the respective telephone number (respective common quick dialling) is automatically transferred to this table from the Mobility Extension IN settings.

Then enter:

Call at day - whether outgoing Mobility Extension should be activated or not if the PBX is in the DAY mode. The setup parameters are "*Never, Immediately, after 1-9 rings"*

Warning: If you select Immediately and the outgoing calls are made via GSM gateway it will take approximately 10 seconds before the external telephone starts ringing (because of the connection method within the GSM network).

Call at night - whether outgoing Mobility Extension should be activated or not if the PBX is in the NIGHT mode. The setup parameters are "*Never, Immediately, after 1-9 rings".*

Warning: If you select Immediately and the outgoing calls are made via GSM gateway it will take approximately 10 seconds before the external telephone starts ringing (because of the connection method within the GSM network).

Call Type - enter, over which bundle of external lines the respective outgoing Mobility Extension should be made. Selected can be only those bundles that do not include analogue external lines.

Wait for trunk - whether the system should wait until the busy line in the respective bundle has been released or not. If all lines in the respective bundle are busy and "Waiting for bundle" is allowed the system waits until the external line in the bundle is released. Otherwise the outgoing call is not made.

Send SMS - The SMS will / will not be sent:

After accepted ME call - Called party answers the call on the ME station = call is established.

Answered ME - Called party does not answer the call on the ME station, but the call was answered on the ralted extension.

Answered call - Called party does not answer the call on any of his stations (ME or certain extension).

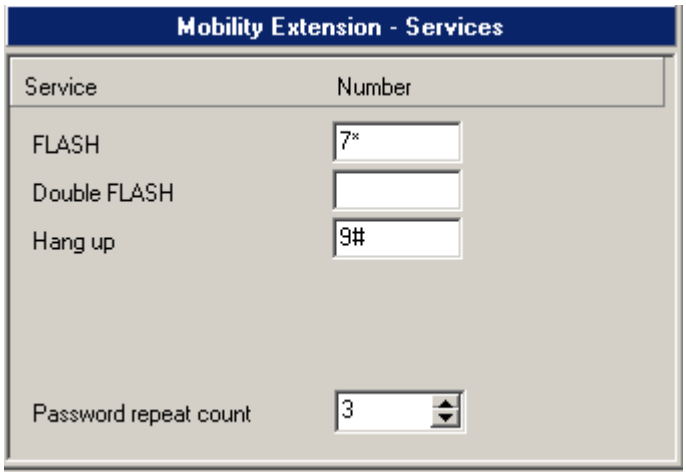
SMS at No Answer – The text is used for SMS if the called party does not answer the call (the missed call) (text is common with text for all outgoing calls thru internal GSM gateway).

SMS after accepted call – The text is used for SMS if the called answers the call (the established call) (text is common with text for all outgoing calls thru internal GSM gateway).

%n = Calling party number
%c = Calling party name

2.10.3. Mobility Extension services

These services are executed based on a command from Mobility Extension. After detecting the DTMF characters entered on the end of the external telephone number this service is simulated in the PBX.



Here enter:

FLASH - DTMF characters, after which FLASH (a short line interruption) is to be generated in the call in order to be able to transfer the call or dial another service.

Double FLASH - DTMF characters, after which FLASH - FLASH (a short line interruption, twice after each other) is to be generated in the call. Not designed for direct use in the PBX.

Hang up - DTMF characters, after which:

- in the case of transferring calls, the transfer is made but the connection of the external telephone number with the external line is not terminated and it is possible to continue dialling (calling VL, calling VTS, calling a service).
- in the case of hesitating to dial (time supervision of dialling the first and the next digit expires – the subscriber hears a congestion tone), the dialling tone is heard again and the PBX is ready for receiving dialling.

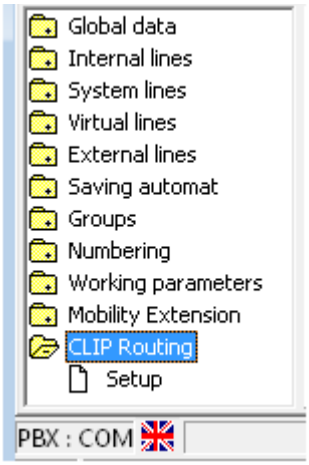
Password repeat count - maximum number of attempts to enter the user password for the line (if required).

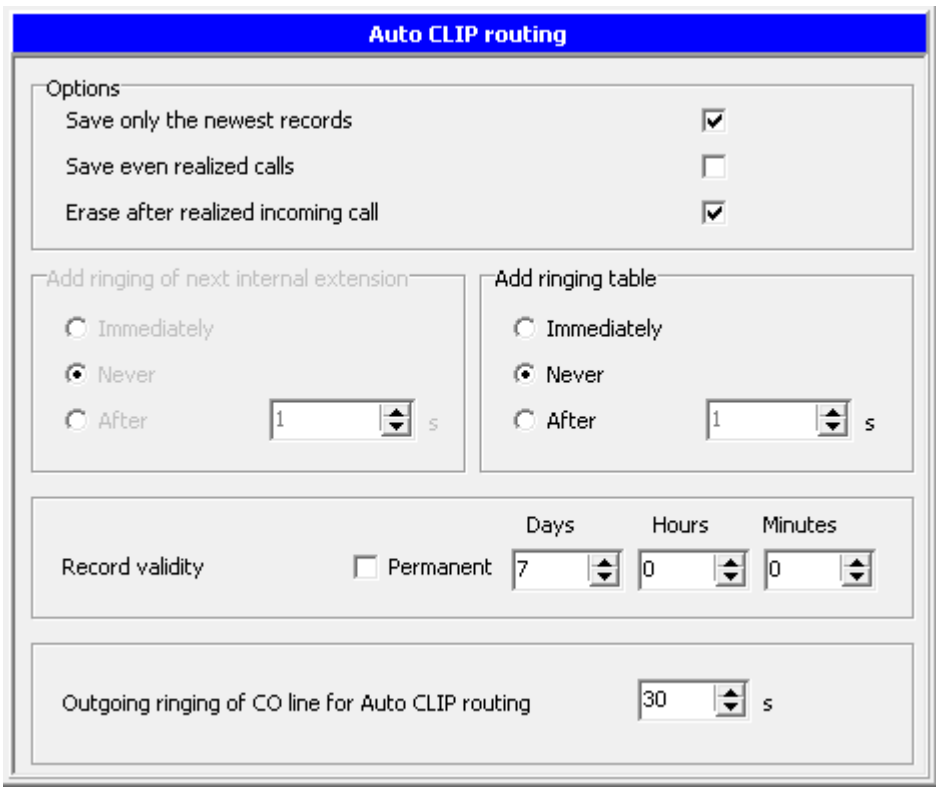
2.11. CLIP Routing

CLIP Routing helps callers from the PSTN to call back the call-establishing subscriber line in the PBX, overriding the ring table rules for the port used.

2.11.1. CLIP Routing - Setting

Set the parameters in the *CLIP Routing/Setup* menu.





Set the following:

- Save latest records only -the CallBack database only includes the last (most recent) calls to the selected PSTN number. If this number was called by multiple subscriber lines, then the last calling station is only saved and the CallBack is routed to this station.
- Save also answered calls -the CallBack database also includes answered calls to the PSTN.
- Erase after answered incoming call -the record is deleted from the CallBack database if a call coming from the record line is answered. Any subsequent call is then governed by the ringing table for the given port.

To define how long a record should be saved in the CallBack database, use the *Record validity* item.
Set the following:

- Permanent -the record is deleted not on a temporary basis but as defined above.
- Days/Hours/Minutes the record is deleted as selected. The maximum record time is 45Days-12Hours-15Minutes.

The CO line Outgoing Call duration - the minimal CO line outgoing call duration (including ringing), needed the call to be counted as a realized (successful) from the point of view of the Auto CLIP Routing function. Longer calls would be saved to the Auto CLIP Routing call history only if „Save even realized calls“ has been selected.

To add a CallBack ringing to a subscriber line according to the available ringing table, use the *Add ringing table* item.
Set the following:

- Immediately –the ringing table is added together with the CallBack ringing to the subscriber line;
- Never -the ringing table is not applied;
- In - the ringing table is added in a certain number of seconds (1-254).

If older records are saved too (regarding multiple PSTN-calling subscriber lines), then enable the *Add ringing to another subscriber line* option. Ringing is routed to the latest record first and only then older records are added as preset.
Set the following:

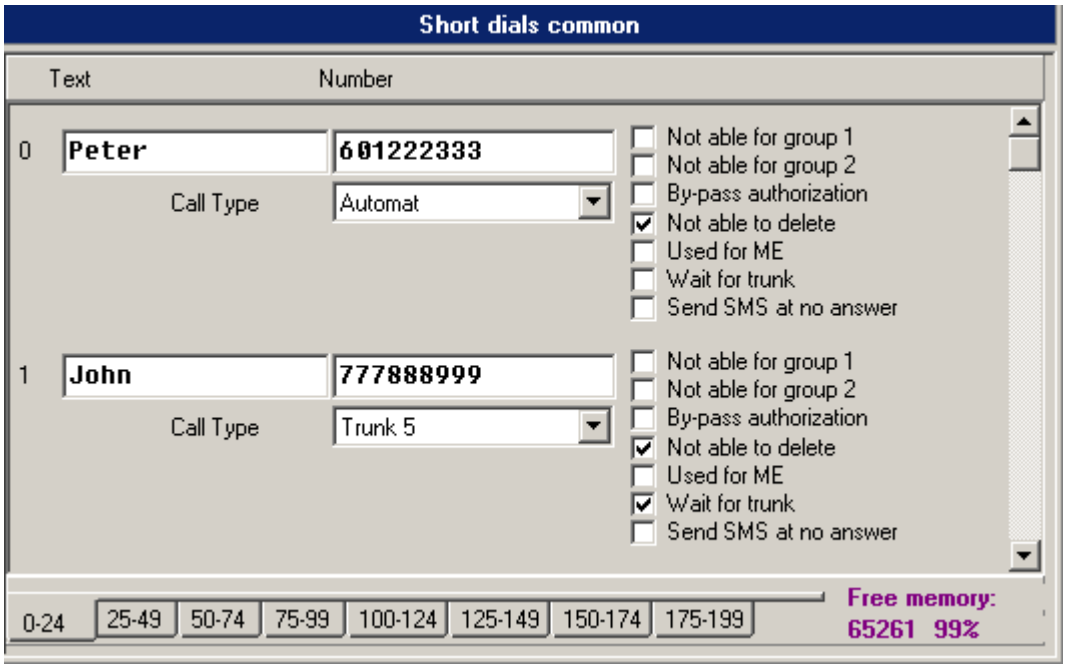
- Immediately -ringing to an earlier record is added together with the CallBack ringing to the subscriber line that created the later record;
- Never –no ringing to an earlier record is applied.
- In -ringing to an earlier record is added to the subscriber line that created the later record in a selected number of seconds (1-254).

The ringing adding options can be combined. *Add ringing to another subscriber line* is applied first, followed by *Add ringing table*.

2.12. Short / Speed - Dialling

2.12.1. Common Abbreviated Dialling

You can enter up to **200 common abbreviated dialling options** into the PBX.

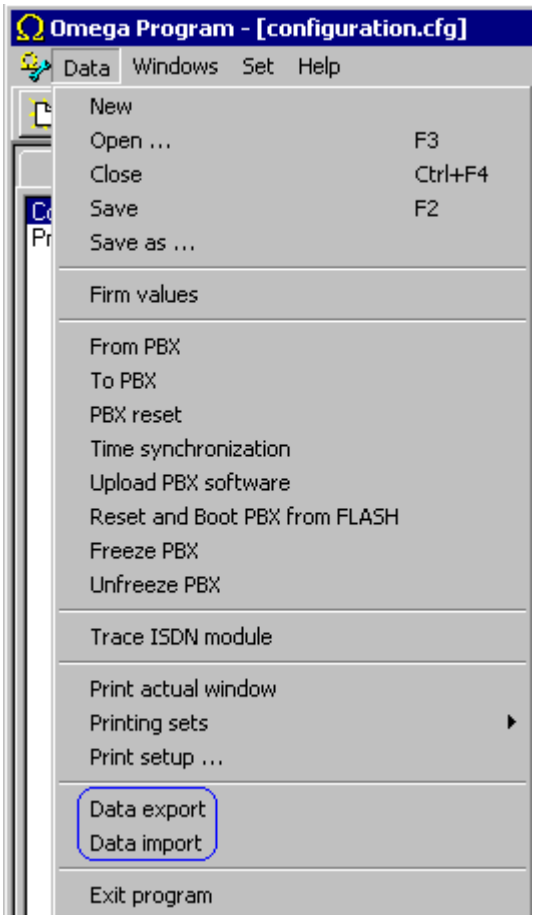


2.12.2. Private Abbreviated Dialling

Key telephone users may enter additional **112 private dialling options** that can be retrieved from the particular key telephone only.

2.13. Export/Import of Abbreviated Dialling

It is possible to Import/Export the Abbreviated Dialling from/to a file. From an active window you may use **Data Export/Data Import** in the Menu ***DATA***.

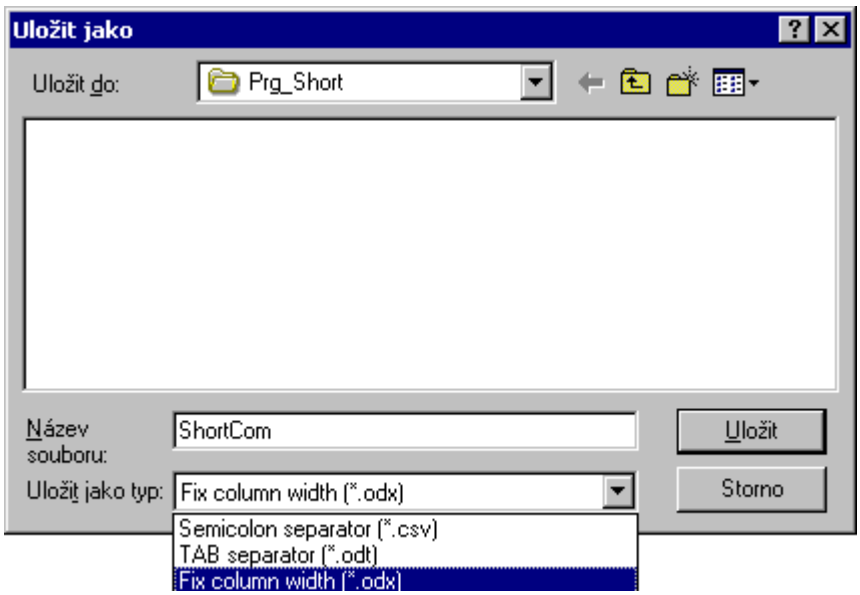


Abbreviated dialling **Export:**

Abbreviated dialling are exported always from the folder, in which you currently are (Common abbreviated dialling, private abbreviated dialling).

2.13.1. Common abbreviated dialling Export

- fill in the Common abbreviated dialling and choose **Export** in Menu Data



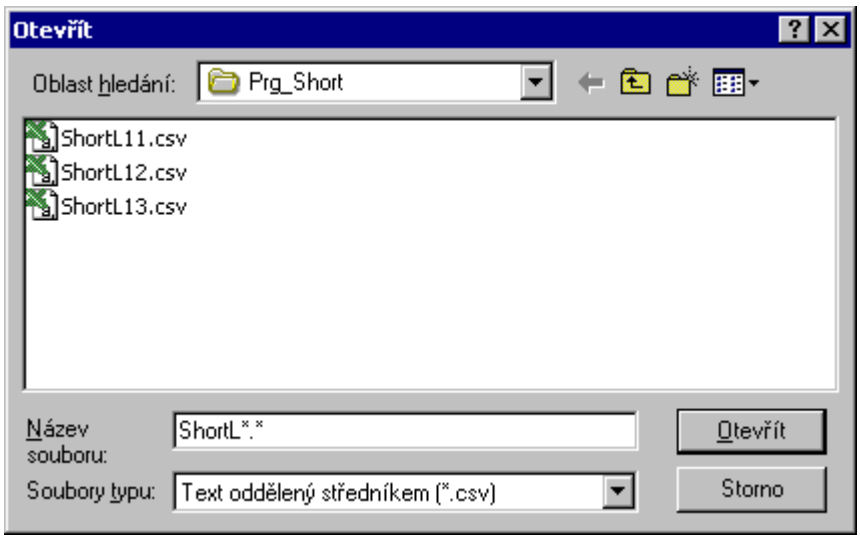
-name of the file is automatically given (ShortCom) with a destination (folder, where OMEGA Lite Program is installed)

- choose the type of columns separation (index number of the dialling, text, number). Separation may be done using semicolon (.csv), tabulator (.odt) or fixed length (.odx).

-click on Save to save the abbreviated dialling

2.13.2. Private abbreviated dialling Export

- fill in the Private abbreviated dialling and choose **Export** in Menu Data



-name of the file is automatically given (ShortLxx) according to the extension number with a destination (folder, where OMEGA Lite Program is installed)

- choose the type of columns separation (index number of the dialling, text, number). Separation may be done using semicolon (.csv), tabulator (.odt) or fixed length (.odx).

-click on Save to save the abbreviated dialling

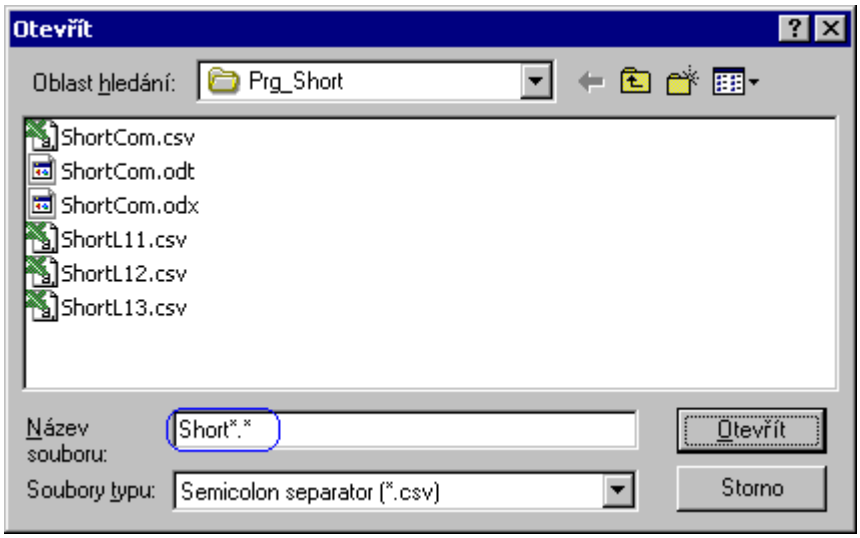
Abbreviated dialling Import

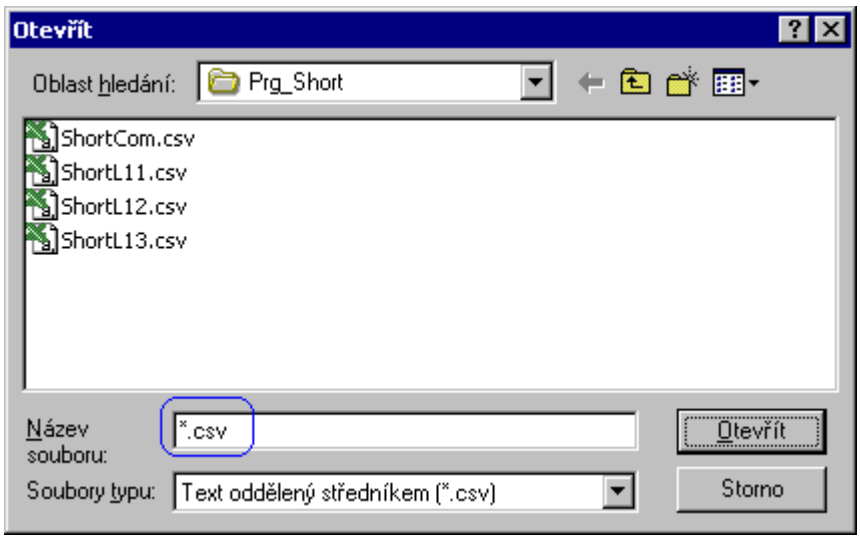
The abbreviated diallings are always imported into the active window (Common abbreviated dialling, Private abbreviated dialling) You may also import them crossways: for example Private abbreviated dialling into Common abbreviated dialling and the other way round. Of course, it must be taken into account, that there could be max. 112 Private abbreviated dialling and 200 Common.

-in the desired section (Private, Common abbreviated dialling) choose command **Import** in the Menu Data.

-choose the desired file (as a first option are offered those files, which belong to the give folder (ShortL.* - private, ShotCom.* - common).

-rest of the files may be found according to the name and extension.

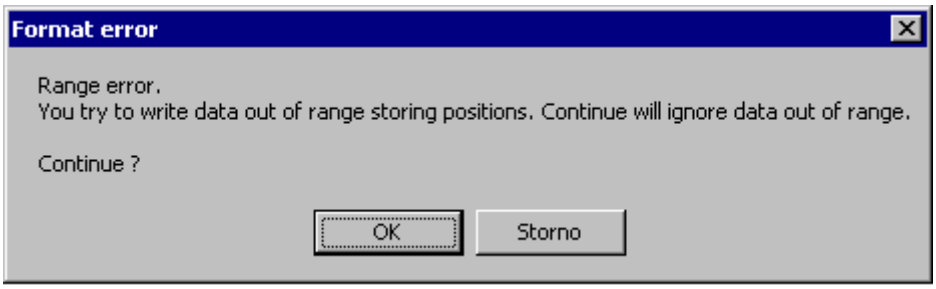




-after clicking Open, the abbreviated dialling is imported.

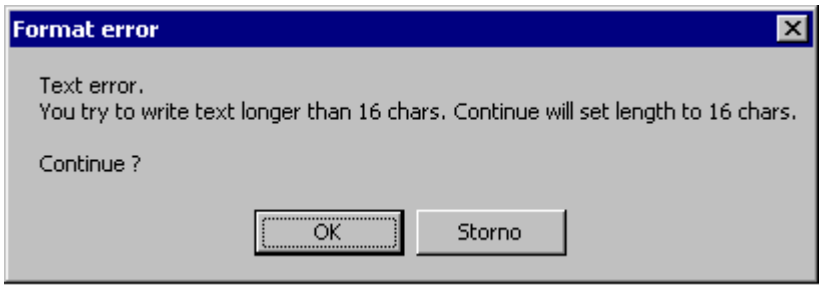
2.13.3. Errors during Abbreviated dialling Import

If a file with abbreviated dialling records is not correctly created or contains

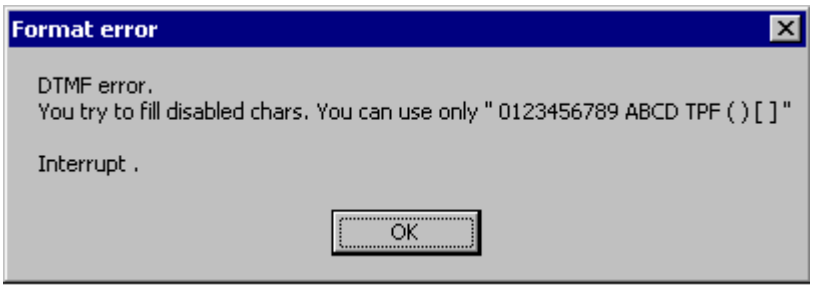

Format error
Range error.
You try to write data out of range storing positions. Continue will ignore data out of range.
Continue ?
OK
Storno

higher number of records, a note is displayed.

-this note means, that the file contains higher number of records, then it is possible to write in the assigned memory space. This can happen if you try to Import Common abbreviated dialling into memory space reserved for Private abbreviated dialling.


Format error
Text error.
You try to write text longer than 16 chars. Continue will set length to 16 chars.
Continue ?
OK
Storno

-in the file, in column "Text" is used more than 16 chars, which is the max. possible value.



- in the file, in column "Number" is used not allowed char. Allowed chars are only "0123456789 ABCD FPT ( ) [ ]".

2.13.4. Creating of Abbreviated dialling file

Abbreviated dialling files have following format and separator:

- semicolon (extension *.scv)
- tabulator (extension *.odt)
- fixed length (extension *.odx)

2.14. REMOTE SUPERVISION

2.14.1. What Remote Supervision Provides:

- instantaneous response to client needs
- *2N*® - OMEGA Lite PBX programming while keeping parameters of a serial link
- firmware modification
- ISDN module tracing
- time RESET setting
- PBX port status monitoring
- speech bus traffic monitoring
- call billing DATA downloading
- log-system DATA downloading

2.14.2. What You Need

To establish a modem-based remote connection you need:

- a PC with installed OMEGA Lite Program
- a modem (not winmodem) connected to the PC
- a file (YYMM-XXXXX.PBX) with access licence to the *2N*® - OMEGA LITE PBX to be supervised.

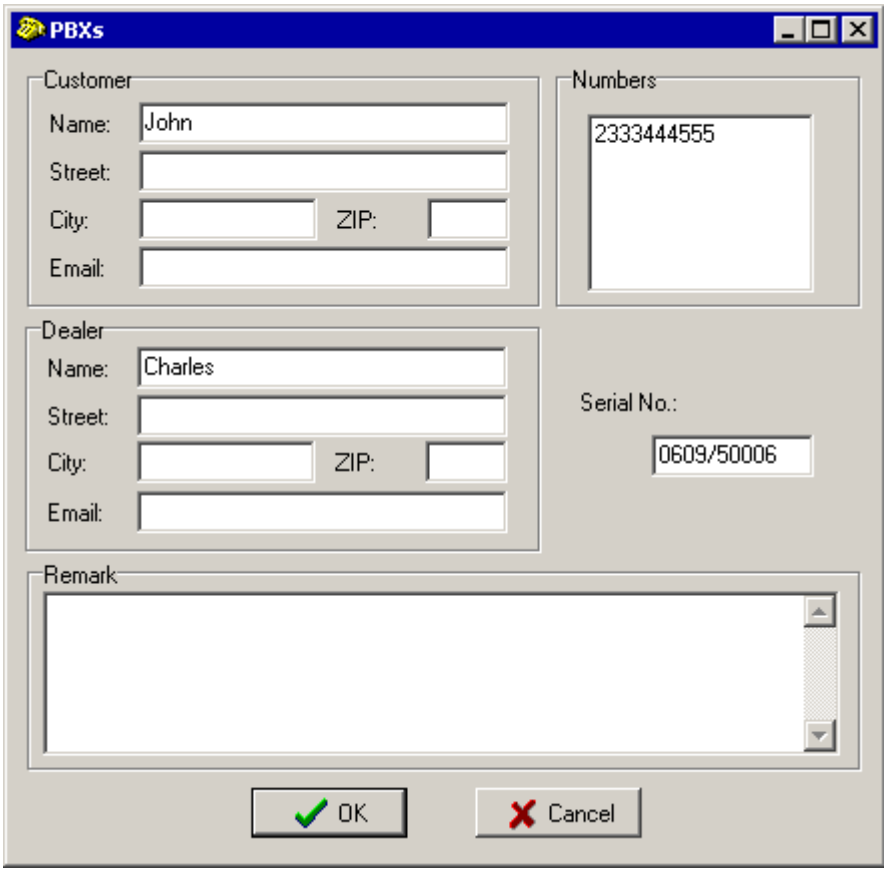
PBX Modem Setting

The internal modem is configured using the OMEGA LITE Program programming tool. The modem is set using AT commands (we do not recommend you to change the factory settings unless absolutely necessary) and the PBX has to be RESET upon every change. The modem change will be apparent 10s after the RESET.

2.14.3. Creating and writing of a gateway remote access code

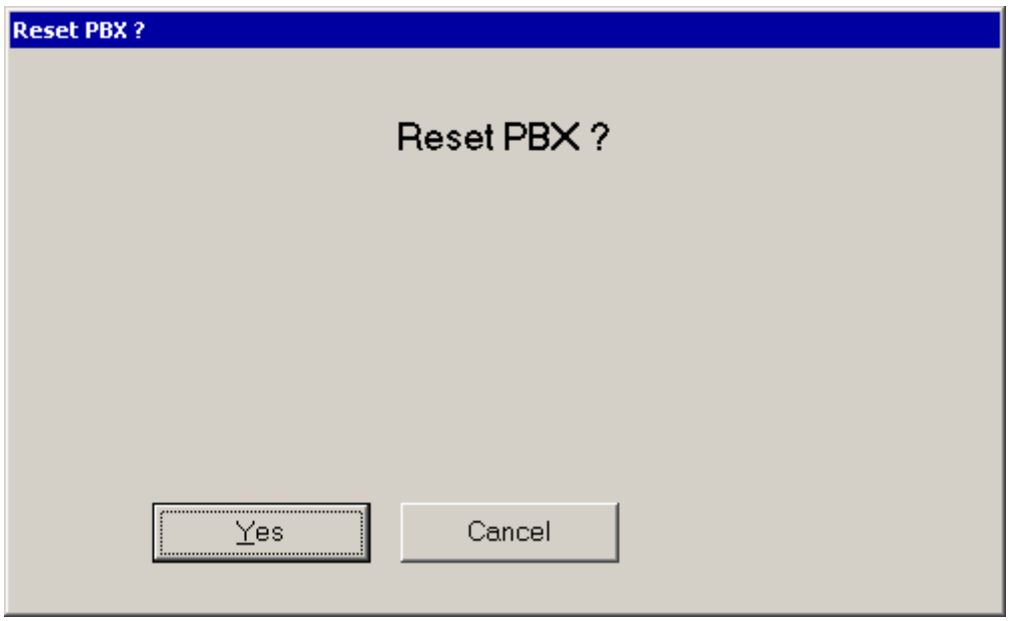
To enable remote control of *2N*® - OMEGA LITE PBX, it is necessary to create an access code and write it into the gateway.

1. Press the „Connect with PBX “ button on the tray.
2. Press button „Create“.



3. you must fill in the PBX *Serial number*.
4. you must fill in a phone number, which will be used for the remote modem connection.
5. you may enter information about customer and dealer.
6. click /OK/ for confirmation.
7. choose the correct code from the offered codes and write it into the PBX by pressing „Store Code“. While writing the code, it is necessary to be connected to the PBX over serial interface.

8. While writing a new code, you will be notified if a code already exists. By pressing Ignore button, code will be overwritten.



Note: Always create access codes matching an existing valid serial number of gateway.

2.14.4. Description of OMEGA Lite Program Menus and Bar Buttons

MENU – WINDOWS

Panel – when the connection with the PBX to be supervised has been established, the statuses of all lines and speech buses are displayed.

Billing data – when billing and diagnostic lines have been read from the file (YYMM-XXXXX.ACC), the billing data (calls) are displayed in the window.

PBX diagnostics - when accounting and diagnostic lines have been read from the file (YYMM-XXXXX.DIA) the diagnostic data (log-system lines) are displayed in the window

Connect to PBX – the window with a directory tree for selecting a file to establish connection with the PBX to be supervised.

Disconnect modem connection – disconnects from the supervised PBX, keeping the selection of the PBX in OMEGA LITE Program.

Recover modem connection – recovers connection with the selected PBX upon Disconnect modem connection.

Cancel connection with PBX – cancels connection with the supervised PBX and selection of the PBX in OMEGA LITE Program.

Set communication – displays the window for communication and connected modem parameter setting.

Download billing and diagnostic lines – downloads call billing and log-system line DATA.

Trace – downloads tracing lines from ISDN.

Download data from PBX – downloads the configuration of the PBX to be supervised.

Save data into PBX – saves the configuration of the PBX to be supervised.

2.14.5. Types of Establishing Connection

It is possible to establish connection with an *2N*® - OMEGA Lite PBX to be supervised:

- automatically; or
- manually.

2.14.6. AUTOMATIC CONNECTION

Settings of PBX To Be Supervised

Automatic connection can be used for a PBX provided that:

1. a DDI to the ISDN line ringing chart is reserved for remote supervision;
2. an ISDN MSN line is reserved for remote supervision;
3. a CO line is reserved for remote supervision;
4. the DISA to the ringing chart is used;
5. the modem connection detection is enabled in DISA.

Example of ringing chart set-up for types 1 – 3:

Interconnect remote supervision	…		
Ring to operator		…	
Wait			0
Hang up			…

Example of ringing chart set-up for types 4 – 5:

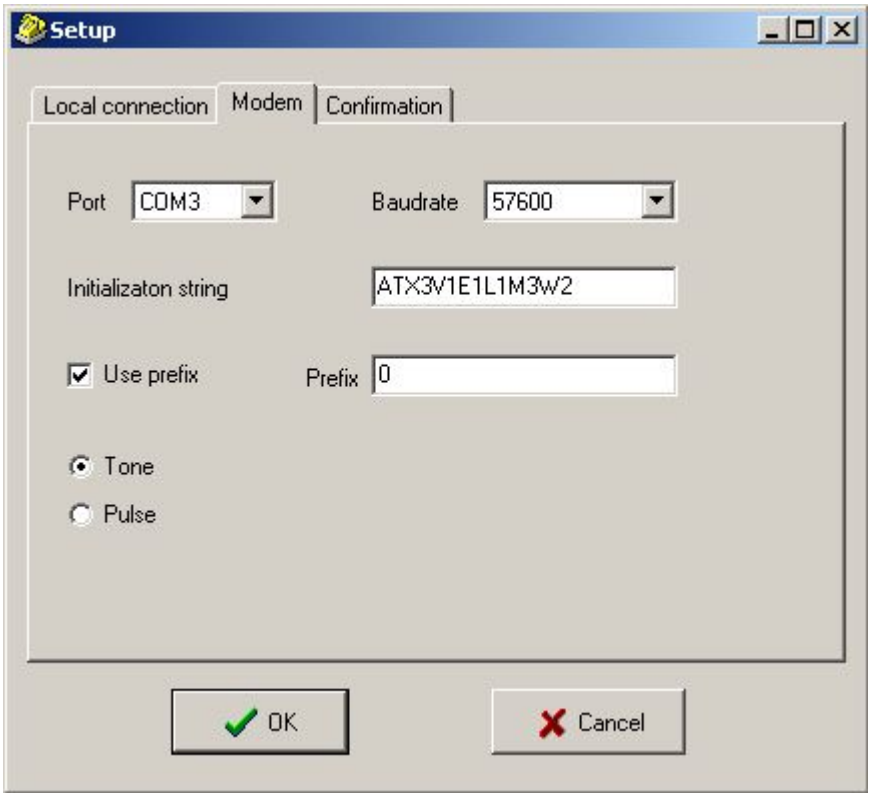
DISA		…
Wait		6
DISA END	…	
Ring to operator	…	
Wait		0
Hang up		…

Note: With remote supervision type 5, you have to set the GLOBAL DATA – MODEM – Modem switch to remote supervision in the programming tool. The DISA tone may not be dialtone permanently.

Establishing Connection

9. Run the OMEGA Lite Program SW.
10. Press the bar button SET COMMUNICATION (‘tools’ symbol) and check and set the modem parameters:

- set the COM port connected to the modem;
- select the communication rate (according to the modem capacities);
- select the use of PREFIX (of the master PBX, set the modem connecting tone transmission (^ or ’up arrow’ symbol (above digit 6) according to the modem type);
- select the CO dialling type;
- confirm the settings by pressing OK.



11. Press the bar button CONNECT TO PBX (three OMEGA LITE symbols in a sequence).
12. Click (using your mouse) on the YYMM-XXXXX.PBX file, which is used for remote supervision.
13. Press the CONNECT MODEM AUTO button.
14. Select the technician responsible for remote supervision, confirm the access password (or select a new technician by pressing the NEW button) and quit the dialog window by clicking on OK.
15. Click on the telephone number used for remote supervision (including DISA number if DISA if used) and press the DIAL button (this dialog window is not displayed if there is only one telephone number).

16. Now the connection is being established.
17. If everything is OK, the following information will be displayed gradually:
- establishing connection;
- connection established;
- remote supervision password confirmed.
18. The following data are displayed on the bottom bar from the left to the right:
- connection type (MOD);
- time of the PBX to be supervised (activity is indicated by a pulsating target);
- production number of the PBX to be supervised (YYMM/XXXXX connected).
11. Go on as if you used a serial link.

2.14.7. MANUAL CONNECTION

Setting of PBX To Be Supervised

Manual connection is used if :

1. No automatic connection can be made.

Example of ringing chart set-up:

Ring to allowed	…	
Ring to operator	…	
Wait		0
Hang up		…

Establishing Connection

1. Run the OMEGA Lite Program SW.
2. Press the bar button SET COMMUNICATION (‘tools’ symbol) and check and set the modem parameters:

- set the COM port connected to the modem;
- select the communication rate (according to the modem capacities);
- select the use of PREFIX (of the master PBX, set the modem connecting tone transmission (^ or ‘up arrow’ symbol (above digit 6) according to the modem type);
- select the CO dialling type;
- confirm the settings by pressing OK.

3. Press the bar button CONNECT TO PBX (three OMEGA LITE symbols in a sequence).
4. Click (using your mouse) on the YYMM-XXXXX.PBX file, which is to be used for remote supervision.
5. Press the CONNECT MODEM MANUAL button.
6. Select the technician responsible for remote supervision, confirm the access password (or select a new technician by pressing the NEW button) and quit the dialog window by clicking on OK.
7. Dial the telephone number to be used for remote supervision on the telephone set connected to the modem, get transferred by the TRANSFER TO REMOTE SUPERVISION (default value # # (the operator dials FLASH ## into the call)) , and then press the ATA button.
8. Now the connection is being established.
9. If everything is OK, the following information will be displayed gradually:

- establishing connection;
- connection established;

- remote supervision password confirmed.

10. The following data are displayed on the bottom bar from the left to the right:
- connection type (MOD);
- time of the PBX to be supervised (activity is indicated by a pulsating target);
- production number of the PBX to be supervised (YYMM/XXXXX connected).

11. Go on as if you used a serial link.

2.14.8. Downloading Data

With the PBX connected, you can download DATA (billing + system messages). To download data, press the bar button DOWNLOAD BILLING AND DIAGN. LINES ($ symbol) of the OMEGA Lite Program tool.

The two DATA types are downloaded together and the course of downloading is monitored graphically by the so-called thermometer.

To display the DATA, select the WIDNOWS – BILLING DATA menu, or PBX DIAGNOSTICS.

Data are displayed from the file (YYMM-XXXXX.ACC, or YYMM-XXXXX.DIA) after being downloaded from the PBX. These files are created upon the first download in the remote supervision file directory (YYMM-XXXXX.PBX) and extended with new data upon re-loading.

2.14.9. Panel

Once the connection has been established, you can display the port status panel of the PBX to be supervised including speech bus configuration by pressing the WINDOWS – PANEL. With the PBX connected, bus 11 and status ‘call’ shall always be displayed for the line used for remote supervision.

2.14.10. Remote Supervision Logistics

A record is made on the remote supervision in progress in the RRMM-XXXXX.LOG file, including when, how and by which technician the connection was established and whether the technician downloaded or saved configuration from/into the PBX.

This file is saved together with the YYMM-XXXXX.PBX file.

2.14.11. Disconnecting Connection

A modem connection can be discontinued (for cost cutting reasons) by pressing the bar button DISCONNECT MODEM CONNECTION (‘pause’ symbol (two perpendicular lines in a green field)).

By pressing this button you discontinue the connection keeping the original connection settings. This means that to recover connection, you just press the bar button RECOVER MODEM CONNECTION (‘replay’ symbol (black arrow in a green field)) and the number will be dialled automatically and the connection will be re-established.

If you used manual connection, you will be invited to dial the telephone number manually.

2.14.12. Cancelling Connection

To cancel the established connection press the bar button CANCEL MODEM CONNECTION WITH PBX (cross in a red field). By pressing this button you cancel the connection and selection of the PBX in the programming tool.

The manufacturer reserves the right, in contrast to the submitted documentation, to make modifications to the product that will improve the product's properties.

Please use the product in accord with the instructions and for the purpose for which it was designed and manufactured.
After the product or its components have come to the end of their lifespan please dispose of them in accordance with the valid legal provisions for environmental protection.